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CONVERSION OF ANALOGUE TO DIGITAL TRANSMISSION CONVERTER

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A project report submitted to the Department of Electrical and Information Engineering in partial fulfillment of the requirements of the degree of BSc. Electrical and Electronic Engineering of the University of Nairobi

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DEDICATION

I would like to dedicate this Project to my family for their moral and financial support during the period of my studies.

ACKNOWLEDGEMENT

I would like to take this opportunity and deeply thank Prof. Maurice Mangoli for his supervision and constant guidance in the accomplishment of the project and providing the means to obtain the respective resources and access relevant in the implementation of this project.

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Thank you all and God bless.

ABSTRACT

In the analog technology, information is translated into electric pulses of varying amplitude while in digital technology; translation of information is into binary format (zero or one) where each bit is representative of two distinct amplitudes.

Analogue transmission involves modulating a continuous beam of charged electromagnetic particles (most commonly radio waves but also microwaves and visible light sent through Fibreoptic cables).

This project entails the analysis of the existing 100kw analogue transmitter paying attention to the respective parameters in modes of transmission that steer the principles of operation including the pros and cons in the transmission evolved. The basic types of transmission based on how they modulate data to combine an input signal with a carrier signal are illustrated, for instance AM and PM. The implementation process is laid out including the process layout and power usage of the transmitter. Relevant features are also laid in line such as bandwidth and noise involvement.

The mystery behind the involvement in a higher power capacity in analogue transmission was elaborated relative to digital transmission i.e. the factor of wide ranges of frequencies and amplitudes explaining more consumption of power.

The digital transmitter is then set on design from the preceding transmitter to transmit binary data of less power capacity. The setback of a narrow area of coverage is combated; for instance, using of boosters in between the stations was depicted.

The discrete messages are either represented by a sequence of pulses by means of a line code (baseband transmission), or by a limited set of continuously varying wave forms (passband transmission), using a digital modulation method. The passband modulation and corresponding demodulation (also known as detection) was carried out by modem equipment.

ABBREVIATIONS AND ACRONYMS

CODEC	Coder Decoder
BCH	Broadcast Channel
IEEE	Institute of Electrical and Electronics Engineering
ADC	Analogue to Digital Converter
DL	Downlink
UL	Uplink
FDM	Frequency-Division Multiplexing
GSM	Global System for Mobile communications
FEC	Forward correction error
MODEM	Modulator Demodulator
DSP	Digital Signal Processing

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CHAPTER 1

INTRODUCTION

1.1: Background to study

Analog signals are continuous in both time and value. Analog signals are used in many systems, although the use of analog signals has declined with the advent of cheap digital signals. All natural signals are Analog in nature.

Analog transmission is a transmission method of conveying voice, data, image, signal or video information using a continuous signal which varies in amplitude, phase, or some other property in proportion to that of a variable. It could be the transfer of an analog source signal, using an analog modulation method such as frequency modulation (FM) or amplitude modulation (AM), or no modulation at all.

Analog transmission can be conveyed in many different fashions e.g. Twisted pair or coax cable, fiber-optic cable, via air, water etc. There are two basic kinds of analog transmission, both based on how they modulate data to combine an input signal with a carrier signal. Usually, this carrier signal is a specific frequency, and data is transmitted through its variations. The two techniques are amplitude modulation (AM), which varies the amplitude of the carrier signal, and frequency modulation (FM), which modulates the frequency of the carrier.

Digital signals are discrete in time and value. Digital signals are signals that are represented by binary numbers, "1" or "0". The 1 and 0 values can correspond to different discrete voltage values, and any signal that doesn't quite fit into the scheme just gets rounded off.

Digital signals are sampled, quantized & encoded version of continuous time signals which they represent. In addition, some techniques also make the signal undergo encryption to make the system more tolerant to the channel.

The process of converting from analog data to digital data is called "sampling". The process of recreating an analog signal from a digital one is called "reconstruction".

1.2 Objectives

To analyze the existing 100kw analogue transmitter to digital transmitter of less power capacity.

1.2.1 Specific Objectives

1. To examine the analogue transmitter and its mode of operation based on the stages in which the signal passes through before transmission

2. The evolution and need of a Digital transmitter in providing a better platform in signal transmission and how power is attenuated in the mode appreciating the cons involved.

3. Investigate how power is reduced in the latter mode and how this can be of benefit.

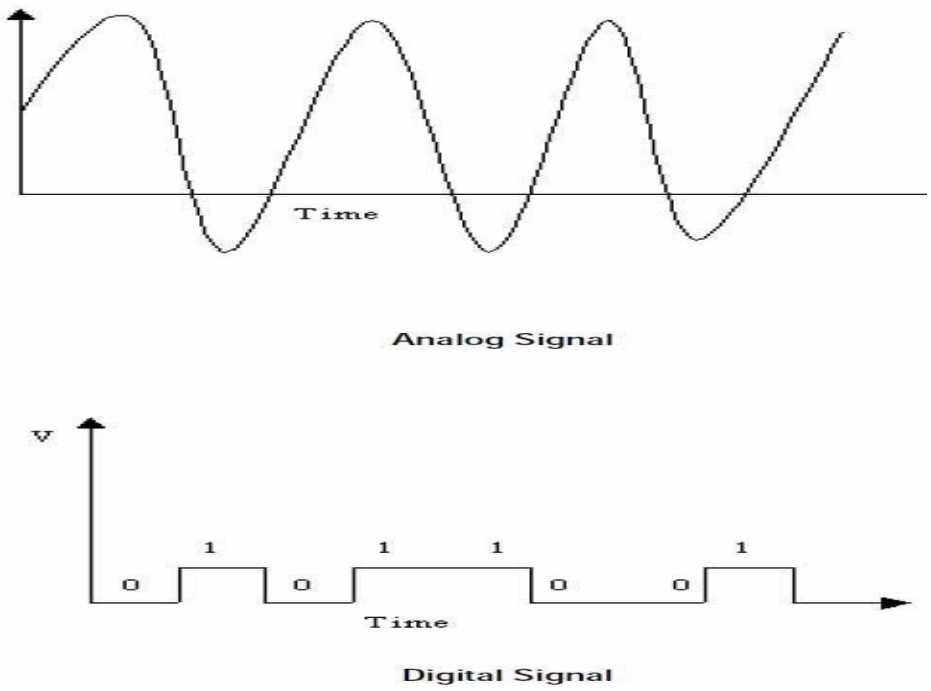


Figure 1.1: Analogue and Digital signals

1.3 REPORT ORGANISATION

This report is organized as follows; the introduction is in chapter one. In chapter two the literature review is presented followed by Methodology which is in chapter three. Chapter four presents the Discussion of the project and finally conclusions and recommendations are discussed in chapter five.

CHAPTER 2

LITERATURE REVIEW

Analog systems occur to be less tolerant to noise, make good use of bandwidth, and are easy to manipulate mathematically. However, analog signals require hardware receivers and transmitters that are designed to perfectly fit the particular transmission.

Digital signals are more tolerant to noise, but digital signals can be completely corrupted in the presence of excess noise. In digital signals, noise could cause a 1 to be interpreted as a 0 and vice versa, which makes the received data different than the original data. The primary benefit of digital signals is that they can be handled by simple, standardized receivers and transmitters, and the signal can be then dealt with in software (which is comparatively cheap to change).

2.1 Signal processing

This is an enabling technology that encompasses the fundamental theory, applications, algorithms, and implementations of processing or transferring information contained in many different physical, symbolic, or abstract formats broadly designated as signals. It uses mathematical, statistical, computational, heuristic, and linguistic representations, formalisms, and techniques for representation, modeling, analysis, synthesis, discovery, recovery, sensing, acquisition, extraction, learning, security, or forensics.

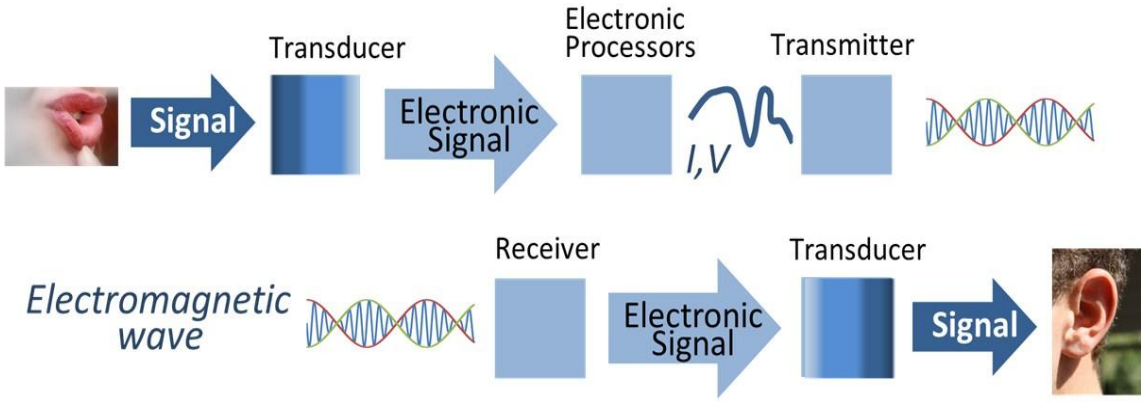


Figure 2.1: Signal processing cycle

2.1.2 Typical devices involved

- Filters: - for example analog (passive or active) or digital (FIR, IIR, frequency domain or stochastic filters, etc.)
- Samplers and Analog-to-digital converters for Signal acquisition and reconstruction, which involves measuring a physical signal, storing or transferring it as digital signal, and possibly later rebuilding the original signal or an approximation thereof.
- Signal compressors
- Digital signal processors (DSPs)

2.1.3 Analog signal processing

This is for signals that have not been digitized, as in legacy radio, telephone, radar, and television systems. This involves linear electronic circuits as well as non-linear ones. The former are, for instance, passive filters, active filters, additive mixers, integrators and delay lines. Nonlinear circuits include compandors, multiplicators (frequency mixers and voltage-controlled amplifiers), voltage-controlled filters, voltage-controlled oscillators and phase-locked loops.

Discrete-time signal processing is for sampled signals, defined only at discrete points in time, and as such is quantized in time, but not in magnitude.

Analog discrete-time signal processing is a technology based on electronic devices such as sample and hold circuits, analog time-division multiplexers, analog delay lines and analog. This technology was a predecessor of digital signal processing, and is still used in advanced processing of gigahertz signals.

The concept of discrete-time signal processing also refers to a theoretical discipline that establishes a mathematical basis for digital signal processing, without taking quantization error into consideration.

2.1.4 Digital signal processing

It is the processing of digitized discrete-time sampled signals. Processing is done by generalpurpose computers or by digital circuits such as ASICs, field-programmable gate arrays or specialized digital signal processors (DSP chips). Typical arithmetical operations include

fixedpoint and floating-point, real-valued and complex-valued, multiplication and addition. Other typical operations supported by the hardware are circular buffers and look-up tables.

Examples of algorithms are the Fast Fourier transform (FFT), finite impulse response (FIR) filter, Infinite impulse response (IIR) filter, and adaptive filters such as the Wiener and Kalman filters.

2.1.5 Nonlinear signal processing

Involves the analysis and processing of signals produced from nonlinear systems and can be in the time, frequency, or spatio-temporal domains. Nonlinear systems can produce highly complex behaviors including bifurcations, chaos, harmonics, and subharmonics which cannot be produced or analyzed using linear methods.

2.2 Sampling

Sampling is the reduction of a continuous signal to a discrete signal. For every T seconds, the sampler reads the current value of the input signal at that exact moment. The sampler then holds that value on the output for T seconds, before taking the next sample. We have a generic input to this system, f (t), and our sampled output will be denoted f*(t). We can then show the following relationship between the two signals: f*(t) = f(0)(u(0) - u(T)) + f(T)(u(T) - u(2T)) + ...

Note that the value of f* at time t = 1.5T = T. This relationship works for any fractional value.

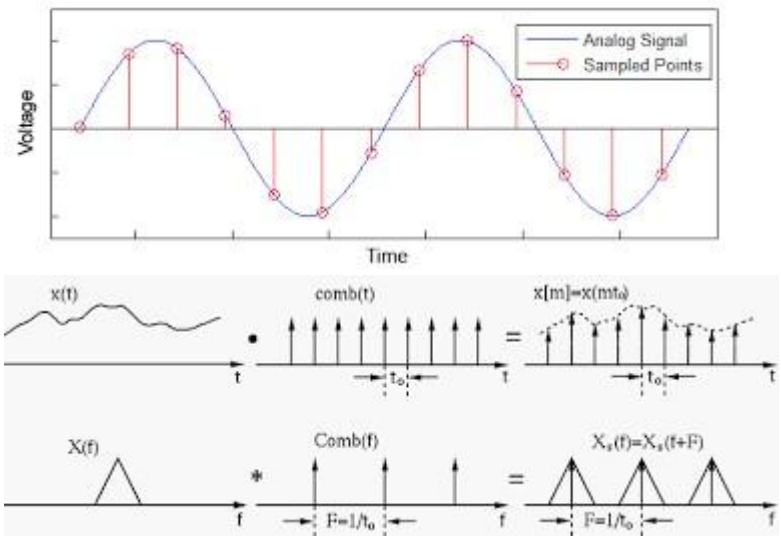


Figure 2.2: Sampling a Signal

2.2.1 Star Transform

Taking the Laplace transform of this infinite sequence will yield us with a special result called the star transform which depends on the sampling time, T, and is different for a single signal, depending on the speed at which the signal is sampled.

$$F^*(s) = \mathcal{L}^*[f(t)] = \sum_{i=0}^{\infty} f(iT)e^{-siT}$$

A sampler is usually denoted on a circuit diagram as being a switch that opens and closes at set intervals. These intervals represent the sampling time, T.

2.2.2 Sampling Time

This is the amount of time between successive samples. Samplers work by reading in an analog waveform, and "catching" the value of that waveform at a particular point in time. This value is then fed into an ADC converter, and a digital sequence is produced.

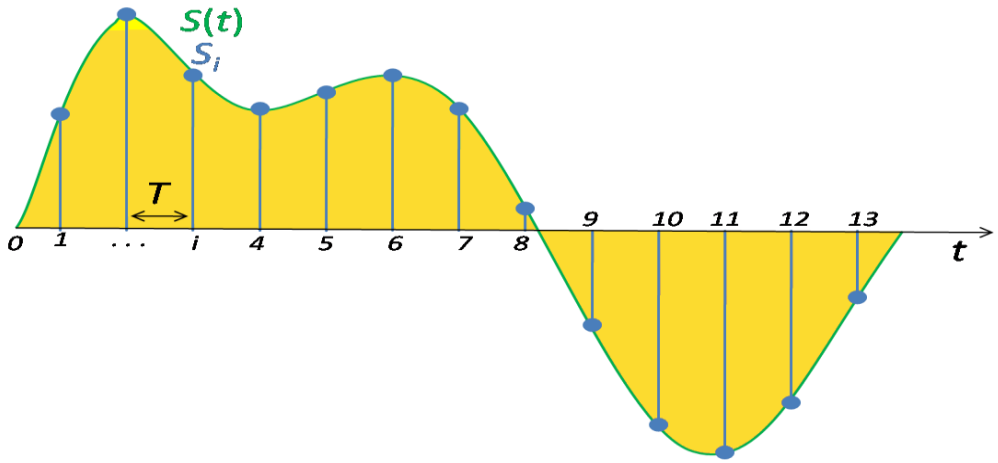


Figure 2.3: Sampling Time

2.2.3 Sampling Delays

Real samplers take a certain amount of time to read the sample, and convert it into a digital representation. This delay can usually be modeled as a delay unit in series with the sampler.

2.2.4 Sampling Jitter

Samplers in real life don't always take a perfect sample exactly at time T, but instead sample "around" the right time. The difference between the ideal sampling time T, and the actual sample time is known as the "Sampling Jitter", or simply the jitter.

Theorem: The sampling theorem states that if we convolute the input function with an impulse, centered at the sampling time T, the output will be the value of the input function at time T. More specifically, our output will be the sample at time T. Here is the definition:

$$\bar{f}(T) = f(t) * \delta(t - T)$$

Where f (T) is the samples value at time T.

2.2.5 Aliasing

If a sampler is only reading in values at particular times, it can become confused if the input frequency is too fast. The resulting problem is called Aliasing, and is a major factor in Sampler design.

When the input signal frequency was faster than half the sampling frequency, the sampled result appears to be a low-frequency wave.

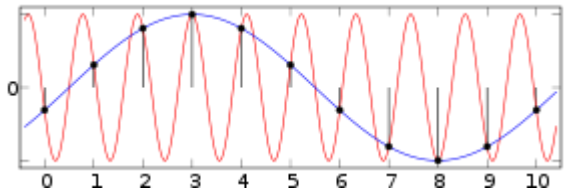


Figure 2.4: Aliasing

2.2.6 Nyquist Sampling Rate

To avoid the problem of aliasing, the Nyquist Sampling Rate should be considered the slowest possible sampling rate. Any slower than the Nyquist sampling rate and the sampler is in danger of producing an aliased signal. The Nyquist sampling rate is two times the highest frequency of the input signal.

The Nyquist sampling rate is a bare minimum, and it is recommended that samplers sample much faster than the minimum. For instance, one common guideline says that it should sample at least 10 times faster than your input signal.

When designing a system, there are 2 ways to prevent ultrasonic sounds (or other unwanted high-frequency noise) from aliasing to a lower frequency, becoming audible noise:

- Adjust the capacitors or other components of the anti-aliasing filter so it blocks all frequencies more than half the sampling rate.
- Adjust the sampling rate to more than twice the frequency of the highest frequency passed by the anti-aliasing filter.

2.2.7 Resolution

The resolution of a sampler is the number of bits that are used to represent each signal. For instance, a 12-bit sampler will output 12 bits of data for every sample. This means that there are 2^{12} possible digital values that each sample can be converted to. In general, the more bits of resolution, the better (more faithful) the digital signal will be to the original. The resolution, n, is related to the number of steps, m, by the following formula:

2.2.8 Unipolar and Bipolar

Samplers come in two basic varieties: unipolar and bipolar. Basically, unipolar samplers only take positive values, and only output unsigned digital values. Bipolar converters can take positive and negative values, and output signed digital values. It is important to note that bipolar converters are generally symmetrical, that is that they have the same number of bits for expressing negative and positive numbers.

2.2.9 Sample Range

The range of possible samples is dependant on a number of factors, including the signed/unsigned number scheme in use by the converter, the resolution, and the step size.

2.2.10 Step Size

The step size of a sampler is the range of analog values that can be input before a bit is changed in the sampler.

Note however, that bipolar converters are generally symmetric. That is, they have the same amount of range below zero as they do above zero. If we want a converter that goes from -5V to +25V, we are going to need to get a converter that can handle from -25V to +25V, which mean we are wasting at least 2/5ths of the possible range of the device.

2.2.11 Bitrate

The number of bits created per sample, times the sampling frequency, gives us the rate at which we are producing data bits. This rate is called the bitrate, and is frequently denoted as r_b, or simply r.

If we have a sampling time of T seconds, then the bitrate and the resolution are related as such:

$$r = \frac{n}{T}$$

Where r is measured in units of bits/second, T is measured in seconds, and n is measured in bits.

2.2.12 Bandwidth

Bandwidth, denoted with a W, is the frequency range needed to transmit an analog or digital signal. Bandwidth is related to the bitrate as follows:

$$W=2r_b$$

This is for a bare, unmodulated bit stream. This value can change depending on what modulation scheme is used, if any.

2.2.13 Down Sampling

There are occasions when the sampler is producing samples too fast, or too slow for the rest of your circuit. When the sampler is producing too many samples, we need to remove some through a process called Down-Sampling. In a down-sampler, certain samples are removed from the digital signal, and the remainder of the samples may be altered to appear more "spread out".

Down-sampling is usually performed according to a fractional rule. An example would be a 2:1 down-sampler, which removes every second sample to decrease the bitrate in half.

2.2.15 Up Sampling

If the sampler isn't producing samples fast enough, we need to create more samples. The process of creating more samples is called Up-Sampling. In the most basic up-sampling scheme, additional samples with a value of zero are added between the existing samples. This method is called "Zero Padding", but other methods, such as interpolation can also be used.

2.2.16 Zero Padding

Adding samples with a 0 value in between given samples to increase the bitrate.

2.2.17 Interpolation

Using some mathematical rule to create new samples a between two existing samples n and m, where a = f (n, m).

2.2.18 Linear Interpolation

In linear interpolation, a straight-line is drawn between the two samples on either side of the new sample. The new sample value then is considered to be a point on this straight line, or the average value. This is called linear interpolation, because the new samples will be on this line formed by the old samples.

As an example, consider that we want to double the sample rate by inserting linearlyinterpolated samples between every two existing samples. In a linear system, the value of the new sample a between existing samples n and m would be:

2.2.19 Non-Linear Interpolations

Analog signals rarely have straight lines in them, and therefore linear interpolation doesn't always produce a good approximation. Nonlinear techniques can be used, taking the surrounding points to produce a new point that isn't just an average value. these methods are called "non-linear interpolation", and there are too many of them for us to give a good example of each.

2.2.20 Sampled Signals

To process signals within a computer (Digital Signal Processing) requires that they be sampled periodically and then converted to a digital representation using an Analog to Digital Converter (ADC). To ensure accurate representation the signal must be sampled at a rate which is at least double the highest significant frequency component of the signal. This is known as the Nyquist rate. In addition, the number of discrete levels to which the signal is quantized must also be sufficient to represent variations in the amplitude to the required accuracy.

Most ADCs quantize to 12 or 16 bits which represent 212 = 4096 or 216 = 65536 discrete levels. After the signal has been processed, it is often necessary to generate an analog output. This function is performed by a Digital to Analog Converter (DAC)

The reconstruction process generally involves holding the signal constant (zero order hold) during the period between samples as shown in the following figure. This signal is then cleaned up by passing it through low-pass filter to remove high frequency components generated by the sampling process.

2.2.21: Conversion: Codecs and Modems

- A codec (which is a contraction of coder-decoder) converts analog signals into digital signals. There are different codecs for different purposes. For the PSTN, for example, there are codecs that minimize the number of bits per second required to carry voice digitally through the PSTN. In cellular networks, because of the constraints and available spectrum, a codec needs to compress the voice further, to get the most efficient use of the spectrum. Codecs applied to video communication also require very specific compression techniques to be able to move those high-bandwidth signals over what may be somewhat limited channels today.
- A modem (which is a contraction of modulator-demodulator) is used to infuse digital data onto transmission facilities. Some modems are designed specifically to work with analog voice-grade lines. There are also modems that are designed to work specifically with digital facilities (for example, ISDN modems, and ADSL modems). A modem manipulates the variables of the electromagnetic wave to differentiate between the ones and zeros.

2.3 Modulation

Modulation is the process of conveying a message signal, for example a digital bit stream or an analog audio signal, inside another signal that can be physically transmitted; a process of varying one or more properties of a periodic waveform, called the carrier signal, with a modulating signal that typically contains information to be transmitted.

There are two principal motivating reasons for modulation. Matching the transmission characteristics of the medium, and considerations of power and antenna size, which impact portability. The second is the desire to multiplex, or share, a communication medium among many concurrently active users.

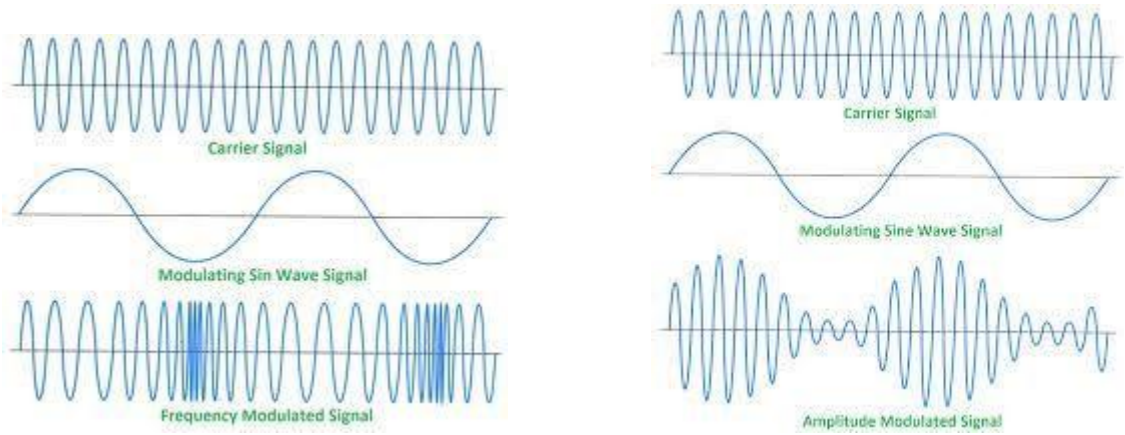


Figure 2.5: Modulation

The aim of digital modulation is to transfer a digital bit stream over an analog bandpass channel, for example over the public switched telephone network (where a bandpass filter limits the frequency range to 300–3400 Hz), or over a limited radio frequency band.

The aim of analog modulation is to transfer an analog baseband (or lowpass) signal, for example an audio signal or TV signal, over an analog bandpass channel at a different frequency, for example over a limited radio frequency band or a cable TV network channel.

Analog and digital modulation facilitate frequency division multiplexing (FDM), where several low pass information signals are transferred simultaneously over the same shared physical medium, using separate passband channels (several different carrier frequencies).

The aim of digital baseband modulation methods, also known as line coding, is to transfer a digital bit stream over a baseband channel, typically a non-filtered copper wire such as a serial bus or a wired local area network.

The aim of pulse modulation methods is to transfer a narrowband analog signal, for example a phone call over a wideband baseband channel or, in some of the schemes, as a bit stream over another digital transmission system.

In music synthesizers, modulation may be used to synthesize waveforms with an extensive overtone spectrum using a small number of oscillators. In this case the carrier frequency is typically in the same order or much lower than the modulating waveform. See for example frequency modulation synthesis or ring modulation synthesis.

2.3.1 Amplitude modulation (AM)

It is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to the waveform being transmitted. That waveform may, for instance, correspond to the sounds to be reproduced by a loudspeaker, or the light intensity of television pixels. This technique contrasts with frequency modulation, in which the frequency of the carrier signal is varied, and phase modulation, in which its phase is varied.

2.3.1.2 Amplitude modulation methods

A low-frequency message signal (top) may be carried by an AM or FM radio wave.

In analog modulation, the modulation is applied continuously in response to the analog information signal. Common analog modulation techniques are:

- ✓ Amplitude modulation (AM) (here the amplitude of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)
- ✓ Double-sideband modulation (DSB)
- ✓ Double-sideband modulation with carrier (DSB-WC) (used on the AM radio broadcasting band)
- ✓ Double-sideband suppressed-carrier transmission (DSB-SC)
- ✓ Double-sideband reduced carrier transmission (DSB-RC)
- ✓ Single-sideband modulation (SSB, or SSB-AM)
- ✓ SSB with carrier (SSB-WC)
- ✓ SSB suppressed carrier modulation (SSB-SC)
- ✓ Vestigial sideband modulation (VSB, or VSB-AM)
- ✓ Quadrature amplitude modulation (QAM)
- ✓ Angle modulation, which is approximately constant envelope
- ✓ Frequency modulation (FM) (here the frequency of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)
- ✓ Phase modulation (PM) (here the phase shift of the carrier signal is varied in accordance with the instantaneous amplitude of the modulating signal)

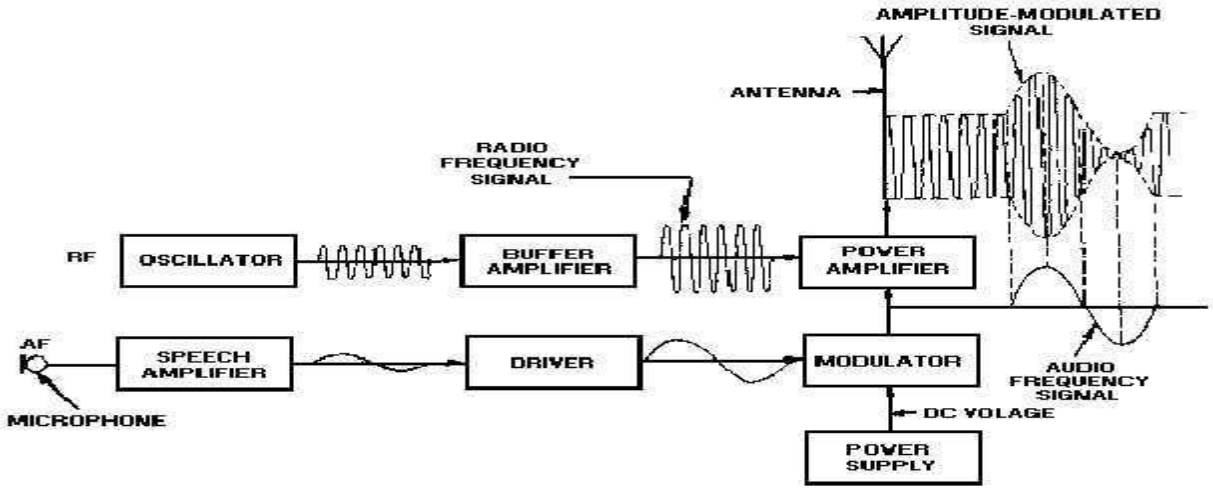


Figure 2.6: Amplitude Modulation

2.3.2 Frequency modulation (FM)

This is the encoding of information in a carrier wave by varying the instantaneous frequency of the wave. (Compare with amplitude modulation, in which the amplitude of the carrier wave varies, while the frequency remains constant.)

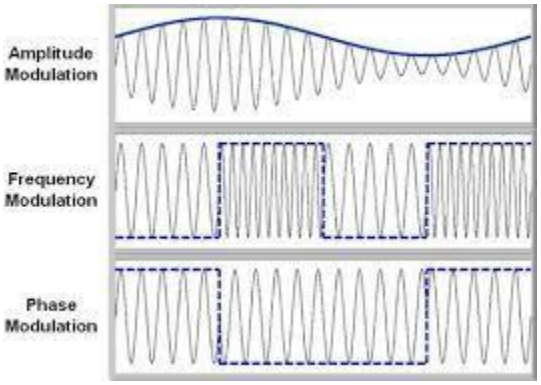


Figure 2.7: Frequency Modulation

In analog signal applications, the difference between the instantaneous and the base frequency of the carrier is directly proportional to the instantaneous value of the input-signal amplitude.

Digital data can be encoded and transmitted via a carrier wave by shifting the carrier's frequency among a predefined set of frequencies—a technique known as frequency-shift keying (FSK). FSK is widely used in modems and fax modems, and can also be used to send Morse code. Radio teletype also uses FSK.

Frequency modulation is used in radio, telemetry, radar, seismic prospecting, and monitoring newborns for seizures via EEG. FM is widely used for broadcasting music and speech, two-way radio systems, magnetic tape-recording systems and some video-transmission systems. In radio systems, frequency modulation with sufficient bandwidth provides an advantage in cancelling naturally-occurring noise.

Frequency modulation is known as phase modulation when the carrier phase modulation is the time integral of the FM signal.

2.3.3 Delta Modulation

The sample values of analog waveforms of real world processes are very often predictable -- i.e. the average change from sample to sample is very small. Hence we can make "educated guess" of what the next sample value depending on the current sample value. Though there is error, it is much less than peak to peak signal range. This concept is used in Predictive coded modulation, where instead of sending the signal, it transmits just the prediction errors. Delta Modulation employs Predictive coded modulation to simplify hardware

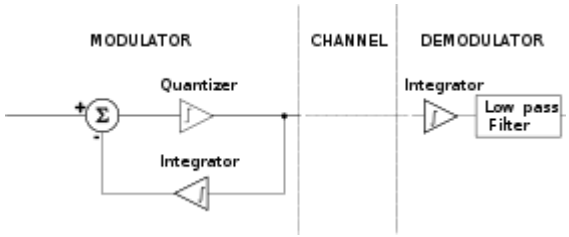


Figure 2.8: Delta Modulation

Delta Modulation is strange in the fact that it attempts to represent an analog signal with a resolution of 1 bit. This is accomplished by successive steps, either up or down, by a preset step size. In delta modulation, we have the stepsize (Δ) that is defined for each sampler, and we have the following rules for output:

If the input signal is higher than the current reference signal, increase the reference by Δ, and output a 1.

If the input signal is lower than the current reference signal, decrease the reference by Δ, and output a 0.

Some benefits of delta modulation are as follows:

1 bit of resolution, and therefore requires very little bandwidth and very little hardware.

No preset upper or lower bounds, so Delta modulation can (theoretically) be used to modulate unbounded signals.

These benefits are countered by the problems of Slope Overload, and Granular Noise, which play an important role when designing a Delta Modulated system.

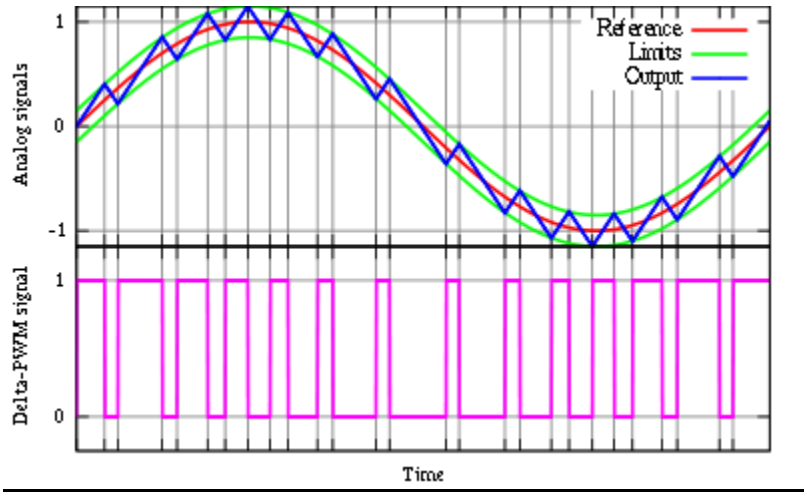


Figure 2.9: Delta Modulated System

2.3.3.1 Slope Overload

If the input signal is rising or falling with a slope larger than Δ/T, where T is the sampling time, we say that the sampler is suffering from Slope Overload. In essence, this means that in a Delta Modulation scheme, we can never have slopes larger than a certain upper limit, and functions that rise or fall at a faster rate, are going to be severely distorted. If the slope of m(n Ts)is greater than the slope of m(n Ts- Ts), then Slope Overload distortion occurs.

2.3.3.2 Granular Noise

A problem with delta modulation is that the output signal must always either increase by a step, or decrease by a step, and cannot stay at a single value. This means that if the input signal is level, the output signal could potentially be oscillatory. That is, the output signal would appear to be a wave, because it would go up and down regularly. This phenomena is called Granular Noise.

When used in ADCs (Analog to Digital Converters), this problem can be solved by internally adding additional bit(s) of resolution that correspond to the value of Δ. This way, the LSBs (Least significant bits) that were added can be ignored in the final conversion result.

2.3.3.3 Delta-Sigma Modulation

A delta-sigma ADC -- also called a sigma-delta ADC -- use the delta modulation technique internally.

2.3.4 Broadcast Signals

Radio communication is typically in the form of AM radio or FM Radio transmissions. The broadcast of a single signal, such as a monophonic audio signal, can be done by straightforward amplitude modulation or frequency modulation. More complex transmissions utilize sidebands arising from the sum and difference frequencies which are produced by superposition of some signal upon the carrier wave. For example, in FM stereo transmission, the sum of left and right channels (L+R) is used to frequency modulate the carrier and a separate subcarrier at 38 kHz is also superimposed on the carrier. That subcarrier is then modulated with a (L-R) or difference signal so that the transmitted signal can be separated into left and right channels for stereo playback. In television transmission, three signals must be sent on the carrier: the audio, picture intensity, and picture chrominance.

This process makes use of two subcarriers. Other transmissions such as satellite TV and long distance telephone transmission make use of multiple subcarriers for the broadcast of multiple signals simultaneously.

2.4 Demodulation

This is the act of extracting the original information-bearing signal from a modulated carrier wave. A demodulator is an electronic circuit (or computer program in a software) that is used to recover the information content from the modulated carrier wave[1] There are many types of modulation so there are many types of demodulators. The signal output from a demodulator may represent sound (an analog audio signal), images (an analog video signal) or binary data (a digital signal).

2.5 Multiplexing

This is a method by which multiple analog message signals or digital data streams are combined into one signal over a shared medium. The aim is to share an expensive resource. For instance, several signals from different media may be carried using one stream channel. The multiplexed

signal is transmitted over a communication channel, which may be a physical transmission medium. The multiplexing divides the capacity of the low-level communication channel into several high-level logical channels, one for each message signal or data stream to be transferred. A reverse process, known as demultiplexing, can extract the original channels on the receiver side.

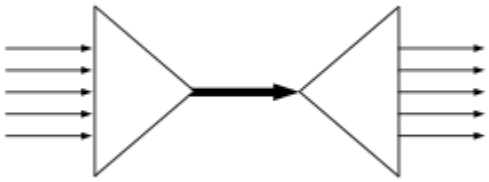


Figure 2.10: Multiplexing

A device that performs the multiplexing is called a multiplexer (MUX), and a device that performs the reverse process is called a demultiplexer (DEMUX or DMX).

Inverse multiplexing (IMUX) has the opposite aim as multiplexing, namely to break one data stream into several streams, transfer them simultaneously over several communication channels, and recreate the original data stream.

2.5.1 Multiplexing Types

Multiple variable bit rate digital bit streams may be transferred efficiently over a single fixed bandwidth channel by means of statistical multiplexing. This is an asynchronous mode timedomain multiplexing which is a form of time-division multiplexing.

Digital bit streams can be transferred over an analog channel by means of code-division multiplexing techniques such as frequency-hopping spread spectrum (FHSS) and directsequence spread spectrum (DSSS).

In wireless communications, multiplexing can also be accomplished through alternating polarization (horizontal/vertical or clockwise/counterclockwise) on each adjacent channel and satellite, or through phased multi-antenna array combined with a multiple-input multipleoutput communications (MIMO) scheme.

2.5.1.1 Space-division multiplexing

In wired communication, space-division multiplexing simply implies different point-to-point wires for different channels. Examples include an analogue stereo audio cable, with one pair of wires for the left channel and another for the right channel, and a multipair telephone cable. Another example is a switched star network such as the analog telephone access network (although inside the telephone exchange or between the exchanges, other multiplexing techniques are typically employed) or a switched Ethernet network. A third example is a mesh network. Wired space-division multiplexing is typically not considered as multiplexing.

In wireless communication, space-division multiplexing is achieved by multiple antenna elements forming a phased array antenna. Examples are multiple-input and multiple-output (MIMO), single-input and multiple-output (SIMO) and multiple-input and single-output (MISO) multiplexing. For example, an IEEE 802.11n wireless router with k number of antennas makes it in principle possible to communicate with k multiplexed channels, each with a peak bit rate of 54 Mbit/s, thus increasing the total peak bit rate with a factor k. Different antennas would give different multi-path propagation (echo) signatures, making it possible for digital signal processing techniques to separate different signals from each other. These techniques may also be utilized for space diversity (improved robustness to fading) or beamforming (improved selectivity) rather than multiplexing.

2.5.1.2 Frequency-division multiplexing

Frequency-division multiplexing (FDM): The spectrum of each input signal is shifted to a distinct frequency range.

Frequency-division multiplexing (FDM) is inherently an analog technology. FDM achieves the combining of several signals into one medium by sending signals in several distinct frequency ranges over a single medium.

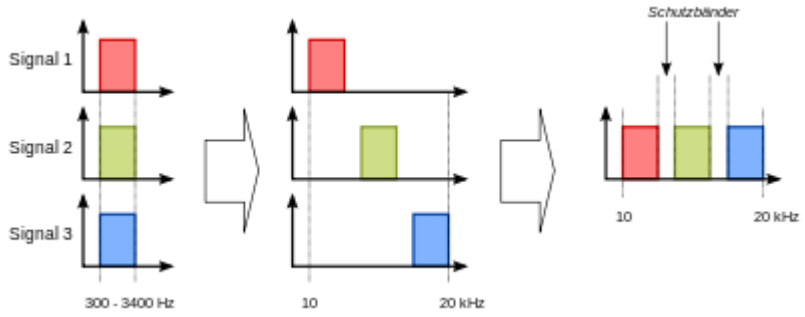


Figure 2.11: Frequency Division Multiplexing

One of FDM's most common applications is the old traditional radio and television broadcasting from terrestrial, mobile or satellite stations, using the natural atmosphere of Earth, or the cable television. Only one cable reaches a customer's residential area, but the service provider can send multiple television channels or signals simultaneously over that cable to all subscribers without interference. Receivers must tune to the appropriate frequency (channel) to access the desired signal.[1]

A variant technology, called wavelength-division multiplexing (WDM) is used in optical communications.

2.5.1.3 Time-division multiplexing

Time-division multiplexing (TDM) is a digital (or in rare cases, analog) technology which uses time, instead of space or frequency, to separate the different data streams. TDM involves sequencing groups of a few bits or bytes from each individual input stream, one after the other, and in such a way that they can be associated with the appropriate receiver. If done sufficiently quickly, the receiving devices will not detect that some of the circuit time was used to serve another logical communication path.

Consider an application requiring four terminals at an airport to reach a central computer. Each terminal communicated at 2400 baud, so rather than acquire four individual circuits to carry such a low-speed transmission, the airline has installed a pair of multiplexers. A pair of 9600 baud modems and one dedicated analog communications circuit from the airport ticket desk back to the airline data center are also installed.

Figure 2.12: Time Division Multiplexing

Some modern web proxy servers (e.g. polipo) use TDM in HTTP pipelining of multiple HTTP transactions onto the same TCP/IP connection.[2]

Carrier sense multiple access and multidrop communication methods are similar to timedivision multiplexing in that multiple data streams are separated by time on the same medium, but because the signals have separate origins instead of being combined into a single signal, are best viewed as channel access methods, rather than a form of multiplexing.

2.5.1.4 Polarization-division multiplexing

Polarization-division multiplexing uses the polarization of electromagnetic radiation to separate orthogonal channels. It is in practical use in both radio and optical communications, particularly in 100 Gbit/s per channel fiber optic transmission systems.

2.5.1.5 Orbital angular momentum multiplexing

Orbital angular momentum multiplexing is a relatively new and experimental technique for multiplexing multiple channels of signals carried using electromagnetic radiation over a single path.[3] It can potentially be used in addition to other physical multiplexing methods to greatly expand the transmission capacity of such systems. As of 2012 it is still in its early research phase, with small-scale laboratory demonstrations of bandwidths of up to 2.5 Tbit/s over a single light path.[4]

2.5.1.6 Code-division multiplexing

Code division multiplexing (CDM) or spread spectrum is a class of techniques where several channels simultaneously share the same frequency spectrum, and this spectral bandwidth is much higher than the bit rate or symbol rate. One form is frequency hopping, another is direct sequence spread spectrum. In the latter case, each channel transmits its bits as a coded channel-specific sequence of pulses called chips. Number of chips per bit, or chips per symbol, is the spreading factor. This coded transmission typically is accomplished by transmitting a unique time-dependent series of short pulses, which are placed within chip times within the larger bit time. All channels, each with a different code, can be transmitted on the same fiber or radio channel or other medium, and asynchronously demultiplexed. Advantages over conventional techniques are that variable bandwidth is possible (just as in statistical

multiplexing), that the wide bandwidth allows poor signal-to-noise ratio according to ShannonHartley theorem, and that multi-path propagation in wireless communication can be combated by rake receivers

2.5.1.7 Forward error correction (FEC)

This is a technique used for controlling errors in data transmission over unreliable or noisy communication channels. The central idea is the sender encodes his message in a redundant way by using an error-correcting code (ECC).

2.6 Microwave and Satellite Systems

Both satellite and ground-based transmissions can use microwaves, which formally are defined as electromagnetic radiation in the wavelength range 0.3 to 0.001 meters, with a frequency between 100 megahertz and 30 gigahertz. This means the waves fall in the spectrum normally used for radar. But both terrestrial and satellite-based microwave transmissions conform to the same physical conditions.

2.6.1 Satellite-Based Transmissions

The C-band uses frequencies between 3.7 and 4.2 GHz, and from 5.9 to 6.4 GHz. The Ku-band satellites use frequencies between 11 and 12 GHz. Both types of communications require ground-based receivers to have a parabolic antenna to receive the signal. The antenna also has to be directed toward the satellite so that it focuses the parabola on the satellite transmission.

2.6.2 Terrestrial Microwave Transmission

Microwave transmission in the atmosphere can only take place when there is a direct line of sight between the sender's and receiver's antenna (point-to-point). This is why microwave transmission towers are speckled with antennas pointing in many directions; they actually point at different microwave transmission towers. The absorption of microwaves in the atmosphere also means that there is very little interference between different microwave towers. Example is the airing of live broadcast from Kenyan parliament

2.6.3 Advantages of Microwave Transmissions

Radio, including microwaves, is a form of energy transmission. Energy transmission at frequencies and wavelengths that are defined as microwaves tend to be absorbed by water molecules. This is why a microwave oven works. For microwave transmission, the water molecules in the atmosphere absorb the transmitted energy.

The effect required for transmission is comparatively low for the amount of data transmitted because of the short distances afforded by the line-of-sight requirement. This is also true for satellites. A satellite can transmit at a relatively low effect, since there is nothing between it and the antenna.

2.6.4 Satellite and Terrestrial Microwave Comparison

Satellite communications only work when there is a line of sight from the communications satellite. So does terrestrial microwave communications. Both require parabolic antennas. This is because apart from the limited frequency bands used by satellite communications, terrestrial and satellite microwave communications are actually using the same technology, and the only difference is the distance between sender and receiver.

2.7 Encoding and Decoding

The process of encoding converts information from a source into symbols for communication or storage. Encoding converts data in one format to another format

Encoding is typically done to utilise one or more of the following advantages:

- Compression of data for more efficient data transfers or storage.
- Improve the quality of a transmission signal - digital encoding is often used to recude the effect of noise and signal attenuation.
- Remove unneeded information relative to the application (digital TV signals consider the quality of human vision and encode the signal accordingly - animals who can see at higher rates than us, such as birds, would be very unimpressed with what they see on the TV!)
- Convert data into a format to communicate with attached peripherals. Encrypt data for security reasons.

Decoding is the reverse process, converting code symbols back into a form that the recipient understands.

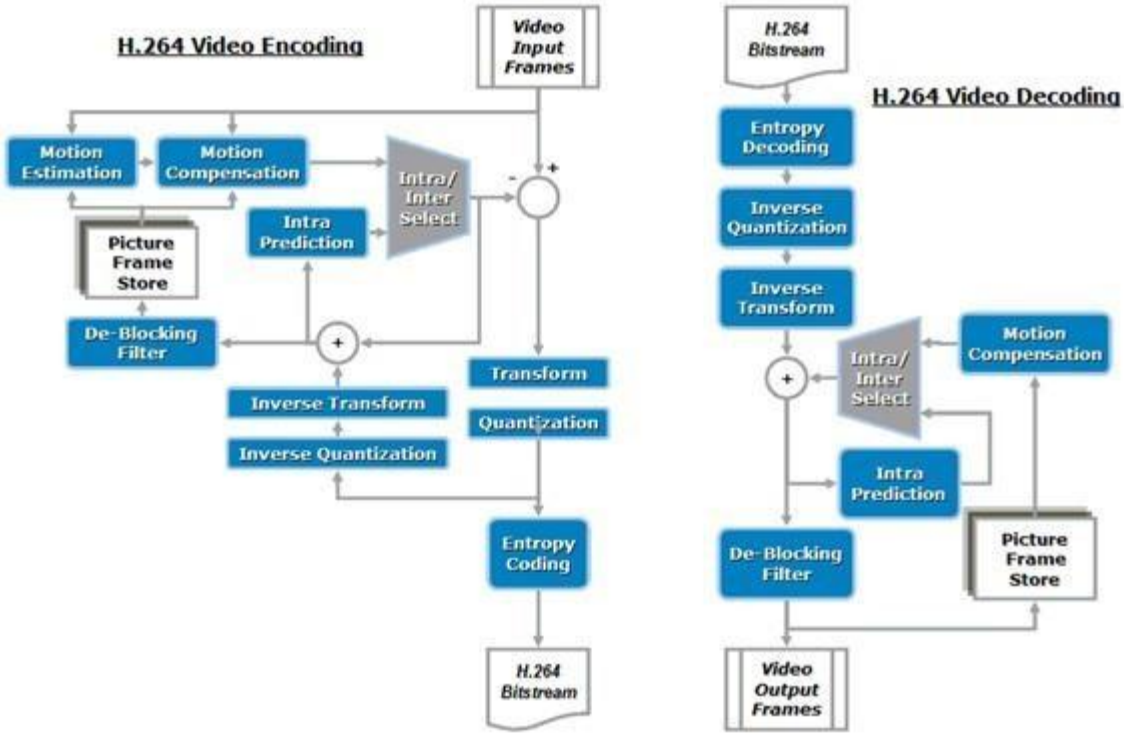


Figure 2.13: Encoding and Decoding

CHAPTER 3

METHODOLOGY

This chapter seeks to illustrate the processes involved in the transmission of a signal from a broadcasting channel. Kenya Broadcasting Channel provided the criteria in doing so.

The KBC Headquarters station contained the various production segments for signal production where processing was then undertaken in the respective control rooms.

Figure 3.1: KBC broadcasting production room

3.1 Signal processing

3.1.1 Sampling

Values of the signal produced were recorded at given points in time. For A/D converters, these points in time are equidistant. The number of samples taken during one second is called the sample rate which are yet analogue values.

In the A/D converters the sampling is carried out by a sample-and-hold buffer. The sample-andhold buffer splits the sample period in a sample time and a hold time. In case of a voltage being sampled, a capacitor is switched to the input line during the sample time. During the hold time it is detached from the line and keeps its voltage.

3.1.2 Quantization

The analog voltage from the sample-and-hold circuit is represented by a fixed number of bits. The input analog voltage is compared to a set of pre-defined voltage levels represented by a unique binary number, and the binary number that corresponds to the level that is closest to the analog voltage is chosen to represent that sample.

This process rounds the analog voltage to the nearest level, which means that the digital representation is an approximation to the analog voltage; for instance; through dual slope or successive approximation

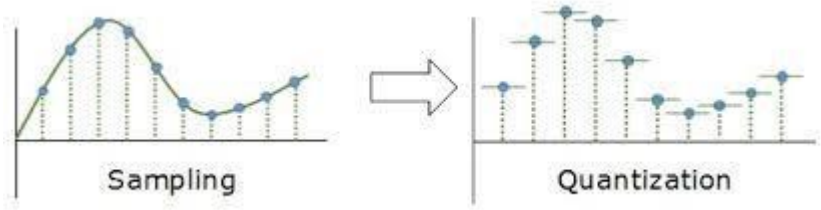


Figure 3.2: Sampled and Quantized Signal

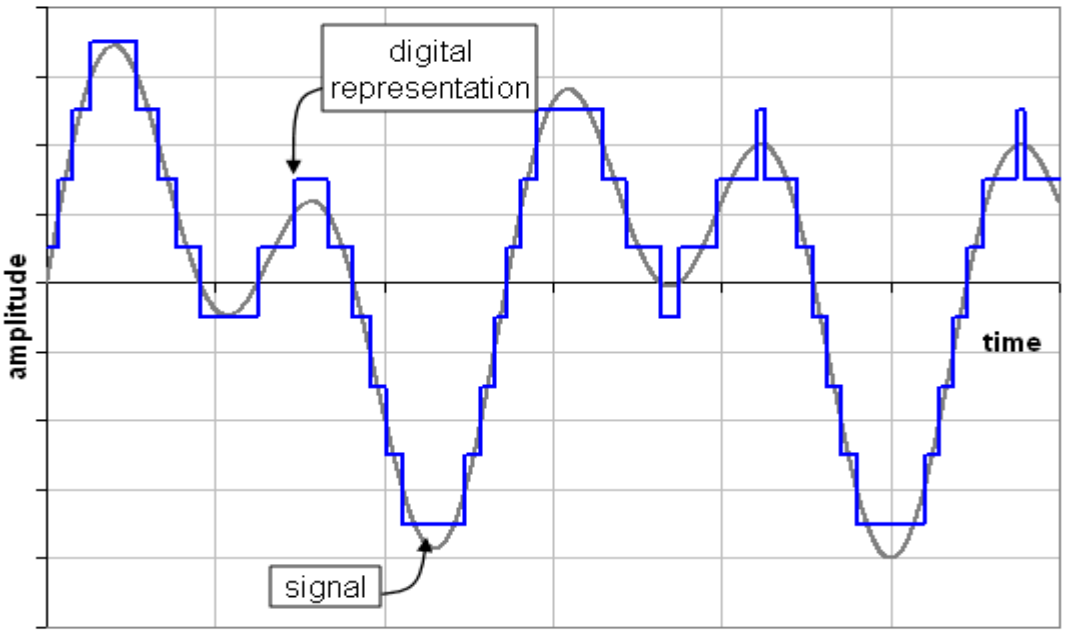


Figure 3.3: Quantization

3.1.3 Reconstruction

Reconstruction is the process of creating an analog voltage from samples. A digital-to-analog converter takes a series of binary numbers and recreates the voltage levels that correspond to that binary number. Then this signal is filtered by a lowpass filter. This process is analogous to interpolating between points on a graph, but it can be shown that under certain conditions the original analog signal can be reconstructed exactly from its samples. The reconstruction is an approximation to the original analog signal.

3.1.4 Aliasing

Due to violation of the Nyquist-Shannon sampling theory, during sampling the base band spectrum of the sampled signal is mirrored to every multifold of the sampling frequency. These mirrored spectra are called alias. The signal spectrum reaches farther than half the sampling frequency base band spectrum and aliases touch each other and the base band spectrum gets superimposed by the first alias spectrum. The easiest way to prevent aliasing is the application of a steep sloped low-pass filter with half the sampling frequency before the conversion. Aliasing can be avoided by keeping Fs>2Fmax.

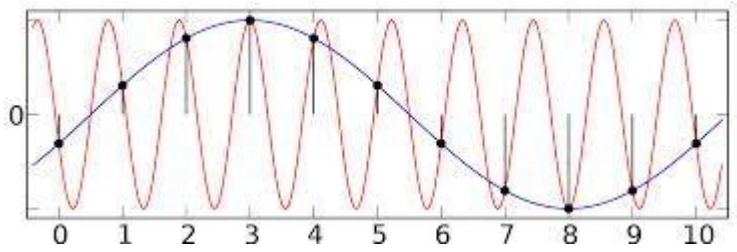


Figure 3.4: Aliasing

3.1.5: Nyquist Sampling Rate

The Nyquist Sampling Rate is the lowest sampling rate that can be used without having aliasing. The sampling rate for an analog signal must be at least two times the bandwidth of the signal. In the sampling controls the sampling rate was set to 44.1 kHz, which is about 10% higher than the Nyquist Sampling Rate to allow cheaper reconstruction filters to be used.

3.1.6: Anti-Aliasing

The sampling rate for an analog signal must be at least two times as high as the highest frequency in the analog signal in order to avoid aliasing. Conversely, for a fixed sampling rate, the highest frequency in the analog signal can be no higher than one half of the sampling rate. Any part of the signal or noise that is higher than one half of the sampling rate will cause aliasing.

In order to avoid this problem, the analog signal gets to be filtered by a lowpass filter prior to being sampled (anti-aliasing filter). Sometimes the reconstruction filter after a digital-to-analog converter is also called an anti-aliasing filter.

3.1.7: Converters

On an incoming analog signal, it is first converted to digital form by an analog-to-digital converter (ADC). The resulting digital signal has two or more levels. Ideally, these levels are always predictable, exact voltages or currents. However, because the incoming signal contains noise, the levels are not always at the standard values. The DSP circuit adjusts the levels so they

are at the correct values. This practically eliminates the noise. The digital signal is then converted back to analog from via a digital-to-analog converter (DAC).

If a received signal is digital, for example computer data, then the ADC and DAC are not necessary. The DSP acts directly on the incoming signal, eliminating irregularities caused by noise, and thereby minimizing the number of errors per unit time.

3.2: DVBT

This is the transmission of digital signal (multiplexed) and using of the frequency spectrum much more efficiently.

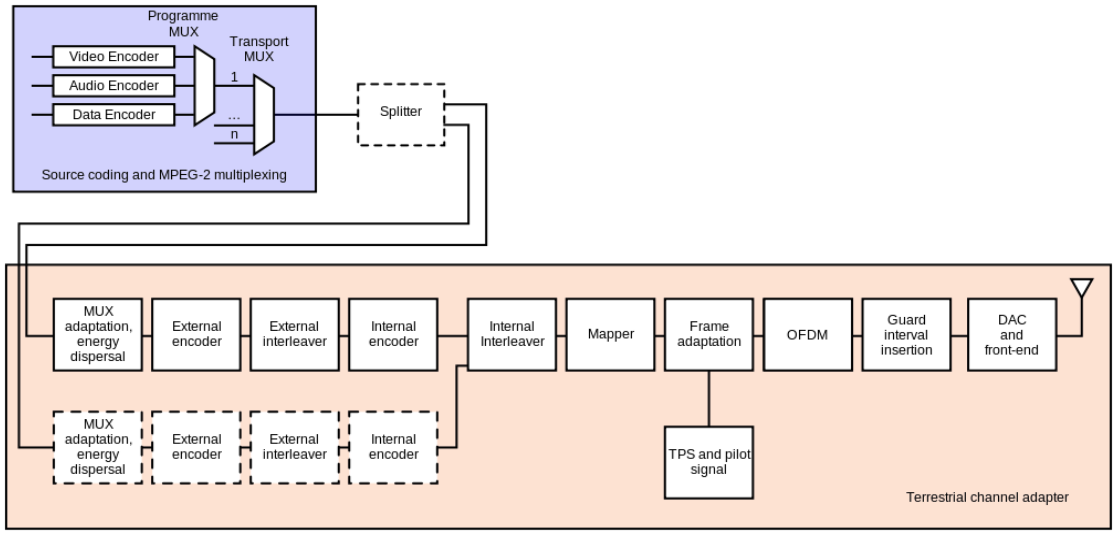


Figure 3.5: DVBT cycle

3.2.1: Source coding and MPEG-2 multiplexing (MUX)

Compressed video, compressed audio and data streams are multiplexed into MPEG program streams (MPEG-PSs). One or more MPEG-PSs are joined together into an MPEG transport stream (MPEG-TS); this is the basic digital stream which is being transmitted and received by TV sets or home Set Top Boxes (STB). Allowed bitrates for the transported data depend on a number of coding and modulation parameters: it can range from about 5 to about 32 Mbit/s

3.2.2: Splitter

Two different MPEG-TS's can be transmitted at the same time, using a technique called Hierarchical Transmission. It may be used to transmit, for example a standard definition SDTV signal and a high definition HDTV signal on the same carrier. Generally, the SDTV signal is more robust than the HDTV one. At the receiver, depending on the quality of the received signal, the STB may be able to decode the HDTV stream or, if signal strength lacks, it can switch to the SDTV one (in this way, all receivers that are in proximity of the transmission site can lock the HDTV signal, whereas all the other ones, even the farthest, may still be able to receive and decode an SDTV signal).

3.2.3: MUX adaptation and energy dispersal

The MPEG-TS is identified as a sequence of data packets, of fixed length (188 bytes). With a technique called energy dispersal, the byte sequence is decorrelated.

3.2.4: External encoder

A first level of error correction is applied to the transmitted data, using a non-binary block code, a Reed-Solomon RS (204, 188) code, allowing the correction of up to a maximum of 8 wrong bytes for each 188-byte packet.

3.2.5: External interleaver

Convolutional interleaving is used to rearrange the transmitted data sequence, in such a way that it becomes more rugged to long sequences of errors.

3.2.6: Internal encoder

A second level of error correction is given by a punctured convolutional code, which is often denoted in STBs menus as FEC (Forward error correction). There are five valid coding rates: 1/2, 2/3, 3/4, 5/6, and 7/8.

3.2.7: Internal interleaver

Data sequence is rearranged again, aiming to reduce the influence of burst errors. This time, a block interleaving technique is adopted, with a pseudo-random assignment scheme (this is really done by two separate interleaving processes, one operating on bits and another one operating on groups of bits).

3.2.8: Mapper

The digital bit sequence is mapped into a base band modulated sequence of complex symbols. There are three valid modulation schemes: QPSK, 16-QAM, 64-QAM.

3.2.9: Frame adaptation

The complex symbols are grouped in blocks of constant length (1512, 3024, or 6048 symbols per block). A frame is generated, 68 blocks long, and a superframe is built by 4 frames.

3.2.10: Pilot and TPS signals

In order to simplify the reception of the signal being transmitted on the terrestrial radio channel, additional signals are inserted in each block. Pilot signals are used during the synchronization and equalization phase, while TPS signals (Transmission Parameters Signaling) send the parameters of the transmitted signal and to unequivocally identify the transmission cell. The receiver must be able to synchronize, equalize, and decode the signal to gain access to the information held by the TPS pilots. Thus, the receiver must know this information beforehand, and the TPS data is only used in special cases, such as changes in the parameters, resynchronizations, etc.

3.2.11: OFDM Modulation

The sequence of blocks is modulated according to the OFDM technique, using 1705 or 6817 carriers (2k or 8k mode, respectively). Increasing the number of carriers does not modify the payload bit rate, which remains constant.

3.2.12: Interval insertion

To decrease receiver complexity, every OFDM block is extended, copying in front of it its own end (cyclic prefix). The width of such guard interval can be 1/32, 1/16, 1/8, or 1/4 that of the original block length. Cyclic prefix is required to operate single frequency networks, where there may exist an ineliminable interference coming from several sites transmitting the same program on the same carrier frequency.

3.2.13: DAC and front-end

The digital signal is transformed into an analogue signal, with a digital-to-analogue converter (DAC), and then modulated to radio frequency (VHF, UHF) by the RF front end. The occupied bandwidth is designed to accommodate each single DVB-T signal into 5, 6, 7, or 8 MHz wide channels. The base band sample rate provided at the DAC input depends on the channel bandwidth

3.3: Processing Techniques in use in audio handling

Audio unprocessed by reverb and delay is metaphorically referred to as "dry", while processed audio is referred to as "wet".

- Echo - to simulate the effect of reverberation in a large hall or cavern, one or several delayed signals are added to the original signal. To be perceived as echo, the delay has to be of order 35 milliseconds or above. Short of actually playing a sound in the desired environment, the effect of echo can be implemented using either digital or analog methods. Analog echo effects are implemented using tape delays and/or spring reverbs. When large numbers of delayed signals are mixed over several seconds, the resulting

sound has the effect of being presented in a large room, and it is more commonly called reverberation or reverb for short.

- Flanger - to create an unusual sound, a delayed signal is added to the original signal with a continuously variable delay (usually smaller than 10 ms). This effect is now done electronically using DSP, but originally the effect was created by playing the same recording on two synchronized tape players, and then mixing the signals together.
- Phaser - another way of creating an unusual sound; the signal is split, a portion is filtered with an all-pass filter to produce a phase-shift, and then the unfiltered and filtered signals are mixed. The phaser effect was originally a simpler implementation of the flanger effect since delays were difficult to implement with analog equipment. Phasers are often used to give a "synthesized" or electronic effect to natural sounds, such as human speech.
- Chorus - a delayed signal is added to the original signal with a constant delay. The delay has to be short in order not to be perceived as echo, but above 5 ms to be audible. If the delay is too short, it will destructively interfere with the un-delayed signal and create a flanging effect..
- Equalization - different frequency bands are attenuated or boosted to produce desired spectral characteristics. Moderate use of equalization (often abbreviated as "EQ") can be used to "fine-tune" the tone quality of a recording; extreme use of equalization, such as heavily cutting a certain frequency can create more unusual effects.
- Filtering - Equalization is a form of filtering. In the general sense, frequency ranges can be emphasized or attenuated using low-pass, high-pass, band-pass or band-stop filters.
- Pitch shift - this effect shifts a signal up or down in pitch. For example, a signal may be shifted an octave up or down. This is usually applied to the entire signal and not to each note separately. Blending the original signal with shifted duplicate(s) can create harmonies from one voice. Another application of pitch shifting is pitch correction. Here a musical signal is tuned to the correct pitch using digital signal processing techniques.
- Time stretching - the complement of pitch shift, that is, the process of changing the speed of an audio signal without affecting its pitch.
- Resonators - emphasize harmonic frequency content on specified frequencies. These may be created from parametric EQs or from delay-based comb-filters.
- Robotic voice effects are used to make an actor's voice sound like a synthesized human voice.
- Synthesizer - generate artificially almost any sound by either imitating natural sounds or creating completely new sounds.
- Modulation- To change the frequency or amplitude of a carrier signal in relation to a predefined signal.

- Compression - the reduction of the dynamic range of a sound to avoid unintentional fluctuation in the dynamics. Level compression is not to be confused with audio data compression, where the amount of data is reduced without affecting the amplitude of the sound it represents.
- 3D audio effects - place sounds outside the stereo basis
- Active noise control- a method for reducing unwanted sound
- Wave field synthesis - a spatial audio rendering technique for the creation of virtual acoustic environments

3.4: Encryption

This includes two components, a digitizer to convert between speech and digital signals and an encryption system to provide confidentiality through the use of Voice Coders (vocoders) to achieve tight bandwidth compression of the signals. Ensures that the sent signal is ready only for transmission avoiding any content alterations.

3.5: Up link

The signal is then send to the transmission sites via uplink or satellite depending on the signal content, distance and its mode of airing.

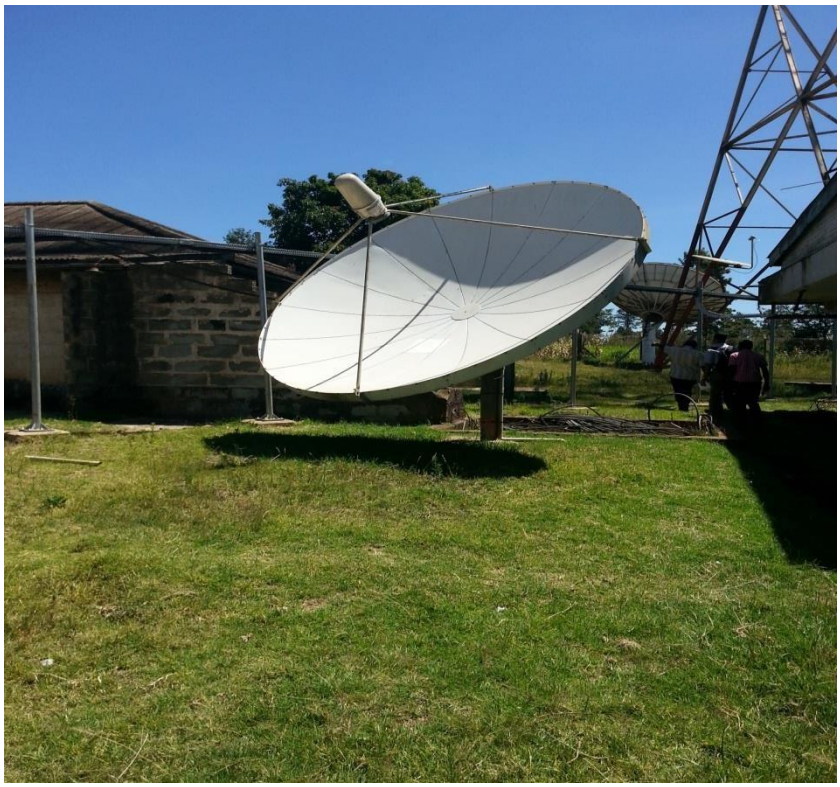


Figure 3.6 Uplink of Signal

3.6: Downlink

The signal is received at the transmission site ready for transmission through the downlink equipment

Figure 3.7: Downlink from the production site

3.7: Analogue FM Transmission (Radio)

The receiver picks up the signal from one the two methods:

- Satellite dish
- Downlink

The 100 kW VHF transmitter is used which uses frequency modulation (FM) to provide high fidelity sound over broadcast radio.

FM radio uses the electrical image of a sound source to modulate the frequency of a carrier wave. At the receiver end in the detection process, that image is stripped back off the carrier and turned back into sound by a loudspeaker.

When information is broadcast from the FM radio station, the electrical image of the sound (taken from a microphone or other program source) is used to modulate the frequency of the carrier wave transmitted from the broadcast antenna of the radio station. This is in contrast to AM radio where the signal is used to modulate the amplitude of the carrier.

The range of mono FM transmission is related to the transmitter's RF power, the antenna gain, and antenna height.

In Nairobi; for instance; the distance for coverage from transmission is approximately 80 Km consisting of sharp depressions along the line.

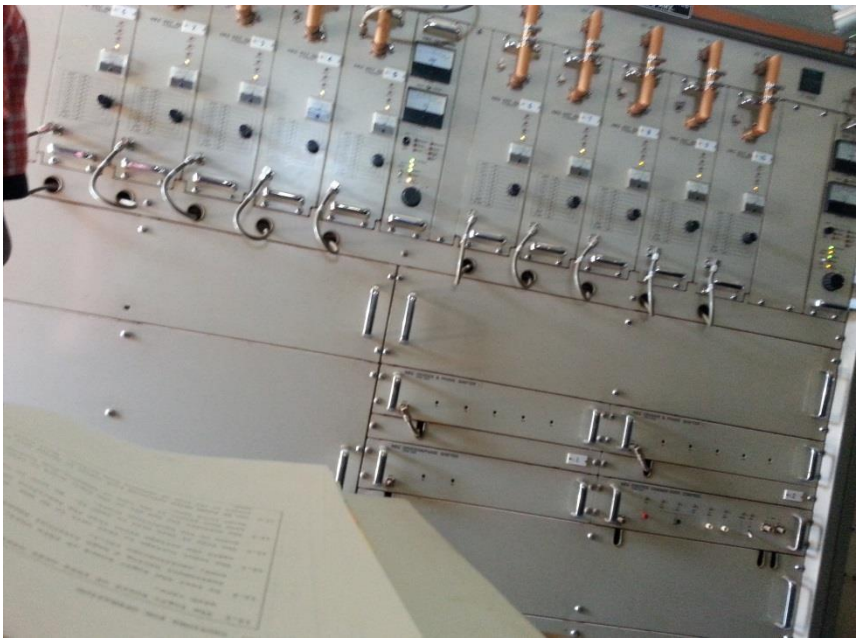


Figure 3.8: A typical FM transmitter control system

The stations for the above system at KBC transmitters include:

1. Kiswahili service - 92.9 MHZ
2. CORO FM - 99.5 MHZ
3. English service - 95.6 MHZ
4. Metro FM - 101.9 MHZ
5. CRI FM (China) - 91.9 FM

The frequency ranges are 88.0 MHz- 108MHz yielding 20 MHz Bandwidth with the transmitter consisting of all frequency ranges due to harmonics.

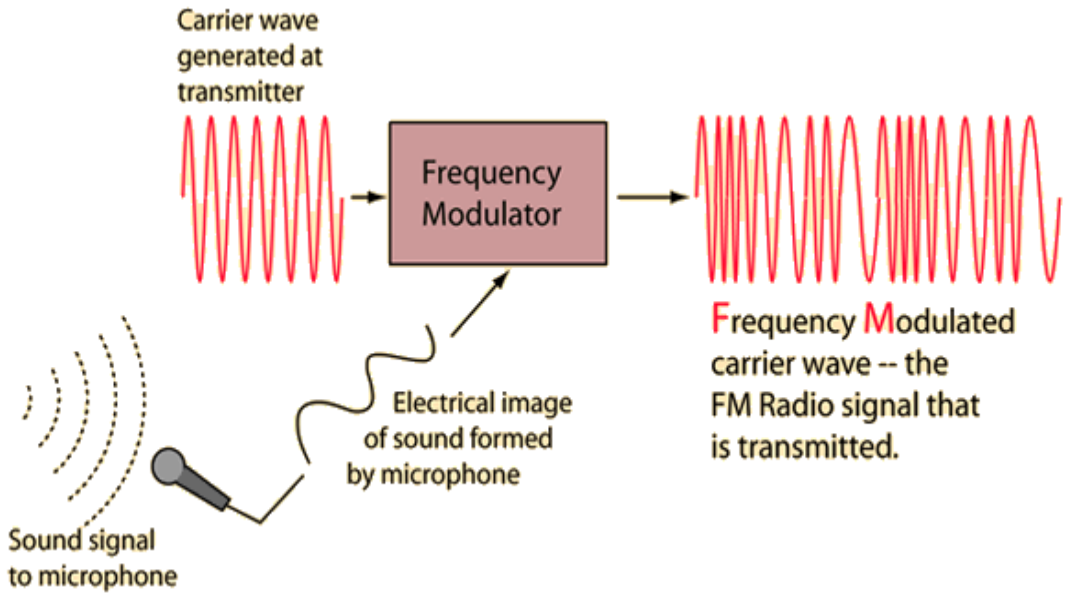

Carrier wave generated at transmitter
Frequency Modulator
Frequency Modulated carrier wave -- the FM Radio signal that is transmitted.
Electrical image of sound formed by microphone
Sound signal to microphone

Figure 3.9: FM Modulation

The stages involved in the transmission are as stated:

3.7.1: Exciter

The signal is passed through the exciter where amplification is undertaken and the phase is synchronized with the carrier wave. The power capacity of this by then is amounted to be 10mW.

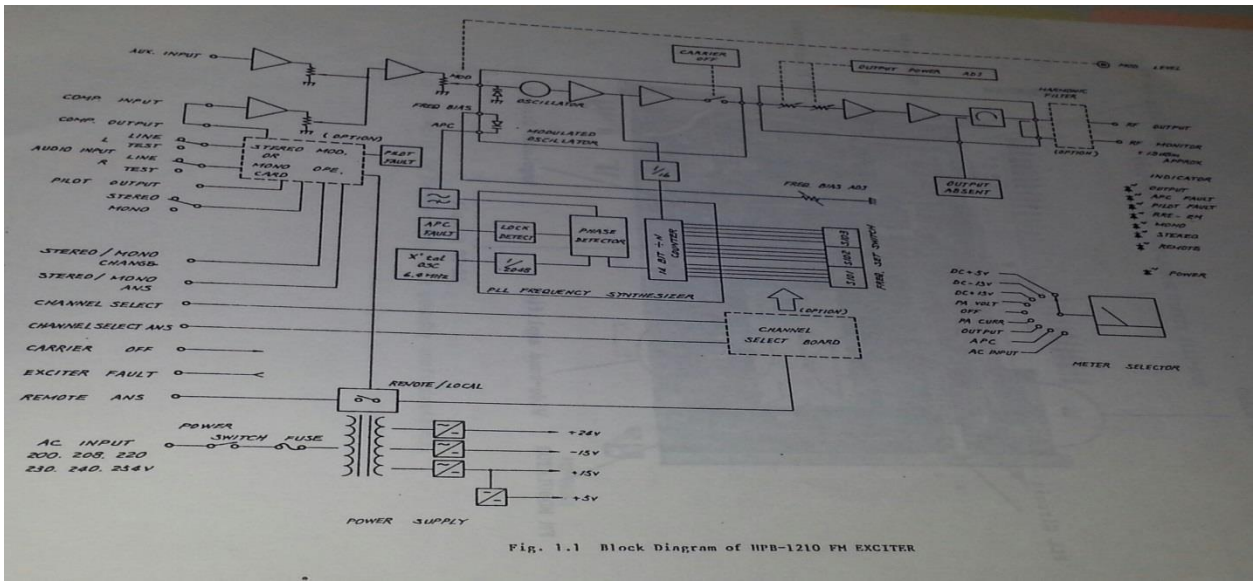


Figure 3.10: Exciter stages

An exciter is used to enhance a signal by dynamic equalization, phase manipulation, harmonic synthesis of (usually) high frequency signals, and through the addition of subtle harmonic distortion. Dynamic equalization involves variation of the equalizer characteristics in the time domain as a function of the input. Due to the varying nature, noise is reduced compared to static equalizers. Harmonic synthesis involves the creation of higher order harmonics from the fundamental frequency signals present in the recording.

As noise is usually more prevalent at higher frequencies, the harmonics are derived from a purer frequency band resulting in clearer highs. Exciters are also used to synthesize harmonics of low frequency signals to simulate deep bass in smaller speakers.

Originally made in valve (tube) based equipment, they are now implemented as part of a digital signal processor, often trying to emulate analogue Exciters. Exciters are mostly found as plugins for sound editing software and in sound enhancement processors.

3.7.2 Divider

Afterwards the signal is advanced to the divider i.e. 2 way or 5 way divider which fits into the respective route.

The RF output of the exciter is next led to the divider and phase shifter divided into five and corrected the phase.

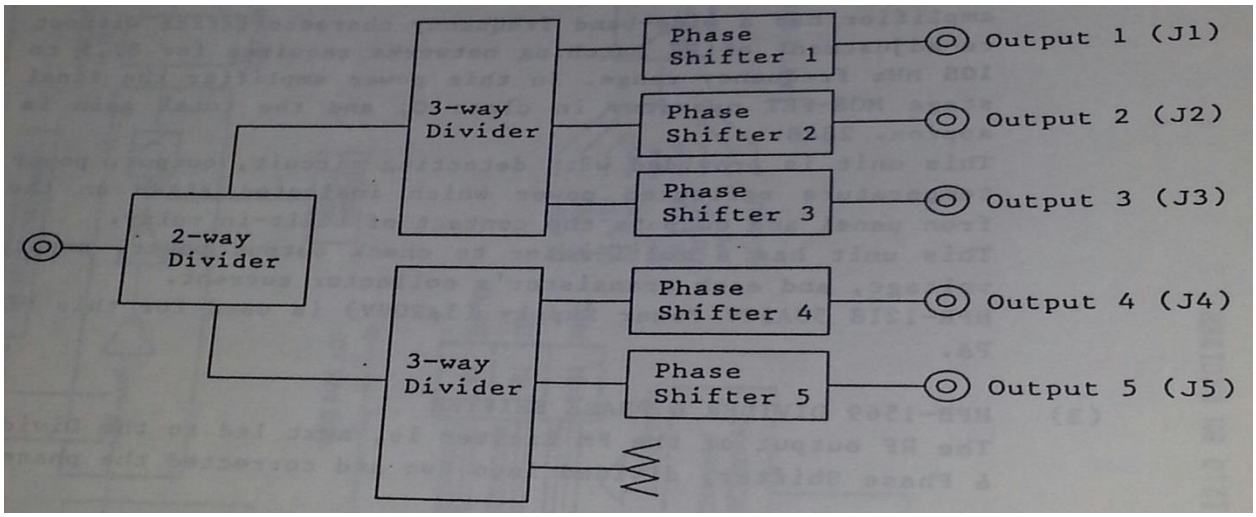


Figure 3.11: 5 way divider block diagram

3.7.3 FET PA

This is used to boost the output power of low power FM broadcast band exciters,

The following is the performance summary:

- 40W min output power
- 88 to 108 MHz frequency range, broadband
- 20dB gain
- +28V DC operation
- High efficiency
- Low component count
- Integrated 7 pole Chebyshev low pass harmonic filter (LPF) Single FET gain stage in class AB

  This design is based on the FET device, with the attendant advantages of:

- High gain
- High efficiency
- Ease of tuning

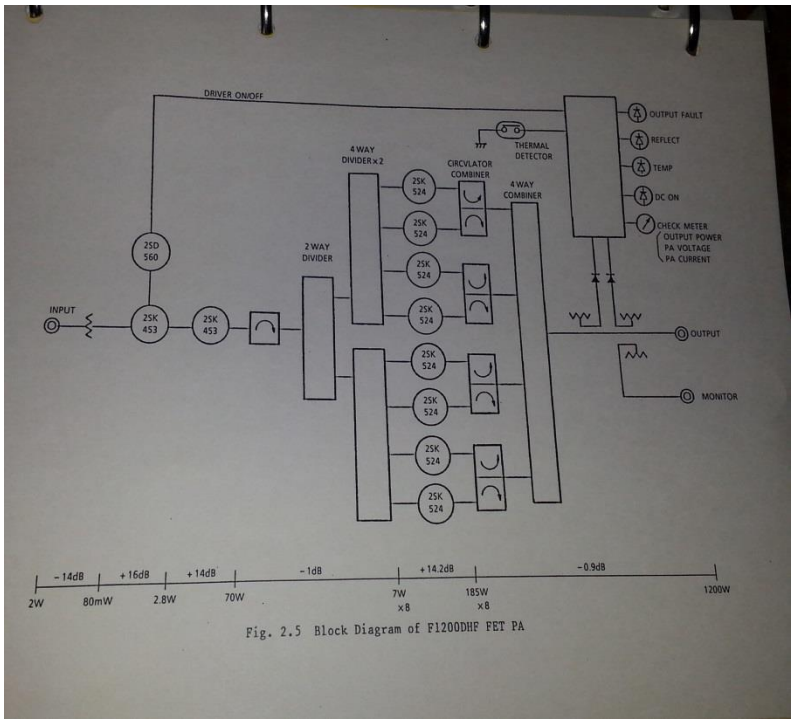


Figure 3.12: Block diagram of FET PA

3.7.4 Filtering

Signal is propagated for filtering to eliminate unwanted frequencies from the receiver signal. While the correct filter settings can significantly improve the visibility of a defect signal, incorrect settings can distort the signal presentation and even eliminate the defect signal completely.

Filtering is applied to the received signal and, therefore, is not directly related to the probe drive frequency as grasped in a time versus signal amplitude display. With this display mode, it is easy to see that the signal shape is dependent on the time or duration that the probe coil is sensing something.

3.7.4.1: Filters Effects

The two standard filters tuned are the 'High Pass Filter' (HPF) and 'Low Pass Filter' (LPF) or a combination high and low pass filter (Band pass filter)

The HPF allows high frequencies to pass and filters out the low frequencies. The HPF is basically filtering out changes in the signal that occur over a significant period of time.

The LPF allows low frequency to pass and filters out the high frequency. In other words, all portions of the signal that change rapidly (have a high slope) are filtered, such as electronic noise.

The gradual (low frequency) changes were first filtered out with a HPF and then high frequency electronic noise was filtered with a LPF to leave a clearly visible flaw indication. Since flaw indication signals are comprised of multiple frequencies, both filters have a tendency to reduce the indication signal strength. Additionally, scan speed must be controlled when using filters. Scan over a flaw too slow and the HPF might filter out the flaw indication. Scan over the flaw too fast and the LPF might eliminate the flaw indication.

3.7.4.2: Filter Settings

If the spectrum of the signal frequency and the signal amplitude or attenuation are plotted, the filter responses can be illustrated in graphical form. The LPF allows only the frequencies in yellow to pass and the HPF only allow those frequencies in the blue area to pass. Therefore, it can be seen that with these settings there are no frequencies that pass (i.e. the frequencies Passed by the LPF are filtered out by the HPF and vice versa).

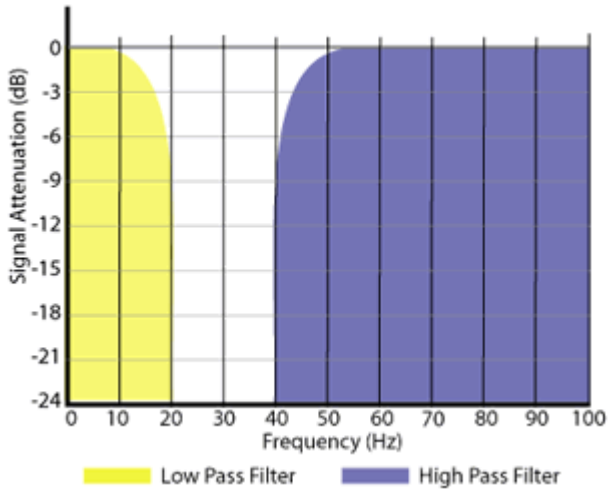


Figure 3.13: Filter effects

To create a window of acceptance for the signals, the filters need to overlap. The area shown in gray is where the two frequencies overlap and the signal is passed. A signal of 30Hz will get through at full amplitude, while a signal of 15Hz will be attenuated by approximately 50%. All frequencies above or below the gray area (the pass band) will be rejected by one of the two filters.

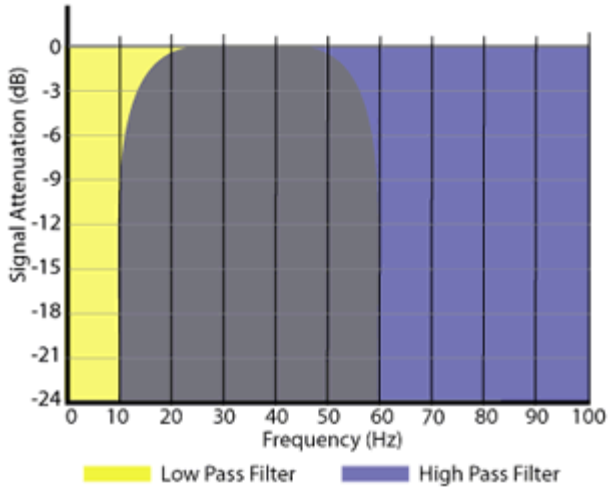


Figure 3.14: Filter effects

3.7.4.3: Use of Filters

The main function of the LPF is to remove high frequency interference noise. This noise can come from a variety of sources including the instrumentation and/or the probe itself. The noise appears as an unstable dot that produces jagged lines on the display as seen in the signal from a surface notch shown in the left image below. Lowering the LPF frequency will remove more of the higher frequencies from the signal and produce a cleaner signal as shown in the center image below. When using a LPF, it should be set to the highest frequency that produces a usable signal. To reduce noise in large surface or ring probes, it may be necessary to use a very low LPF setting (down to 10Hz). The lower the LPF setting, the slower the scanning speed must be and the more closely it must be controlled. The image on the right below shows a signal that has been clipped due to using a scan speed too fast for the selected HPF setting.

The HPF is used to eliminate low frequencies which are produced by slow changes, such as conductivity shift within a material, varying distance to an edge while scanning parallel to it, or out-of-round holes in fastener hole inspection. The HPF is useful when performing automated or semiautomatic scans to keep the signal from wandering too far from the null (balance) point. The most common application for the HPF is the inspection of fastener holes using a rotating scanner. As the scanner rotates at a constant RPM, the HPF can be adjusted to achieve the desired effect.

Use of the HPF when scanning manually is not recommended, as keeping a constant scanning speed is difficult, and the signal deforms and amplitude decreases.

The size of a signal decreases as the scan speed decreases and a flaw indication can be eliminated completely if the scan is not done with sufficient speed. In the images below, it can be seen that a typical response from a surface notch in aluminum without HPF (left image) looks considerably different when the HPF is activated (right image). With the HPF, looping signals with a positive and similar negative deflection are produced on the impedance plane.

3.7.5: Combiner

The signal combiner receives signals from all receiving antennas and provides a combination technique with performance close to optimal (maximal ratio) combiners but without the complexity. The invention provides a base station arrangement which only processes the outputs from antennas which contribute positively to the overall system carrier-to-noise ratio.

Figure 3.15: A typical transmission combiner

3.8: Graceful Degradation

This is the property that enables a system to continue operating properly in the event of the failure of some of its components. The choice of the RF composition and its bias is fundamental in order to guarantee the ON AIR service also during some subsystem fault.

The output power is always present even if there are 3 broken pallets. Obviously the output power will be reduced a percentage obtainable by the formula:

$$GD = \left(\frac{M-N}{M}\right)^2$$

Where M= Total number of amplifiers in parallel N= Failures amplifiers

$$GD_{db} = 20 \text{ Log } \frac{M-N}{M}$$

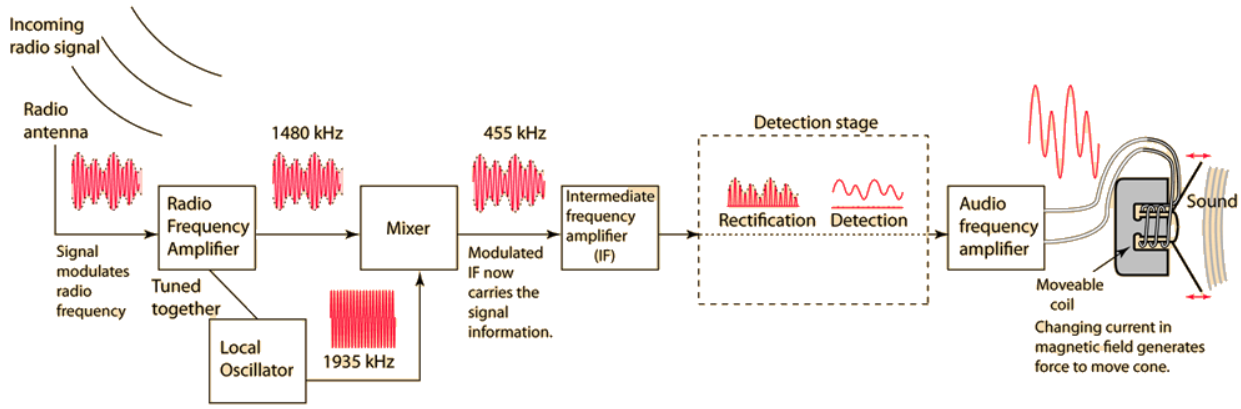


Figure 3.16: A typical FM transmission block diagram

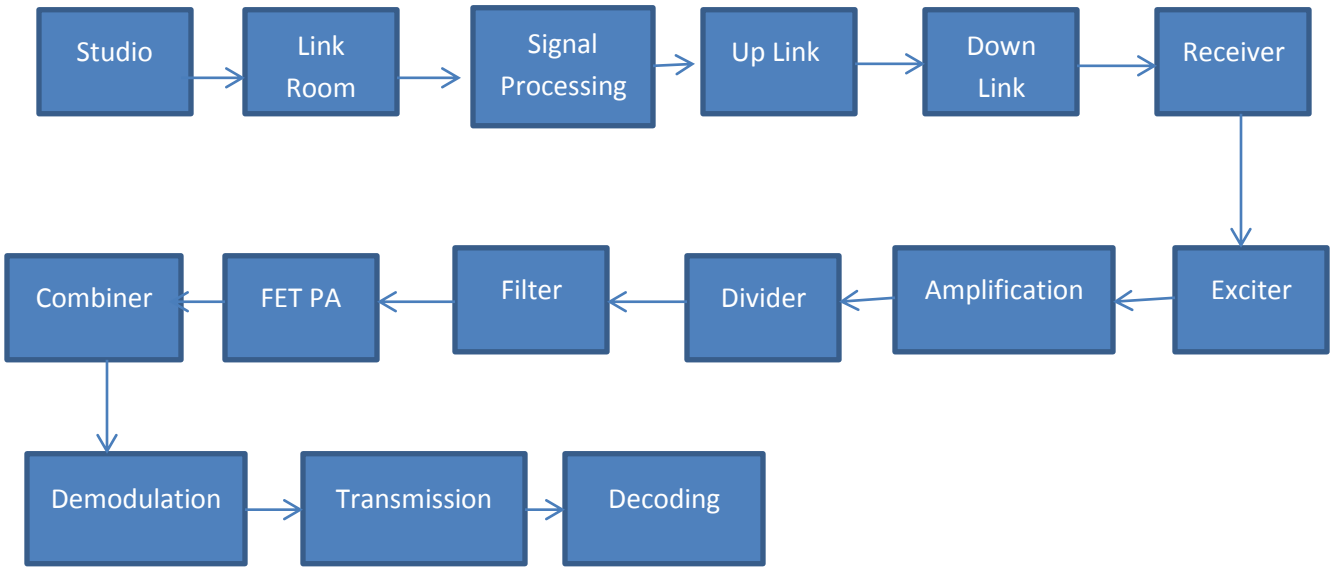


Figure 3.17: The block diagram for FM Transmission

3.9: Analogue Transmission (TV)

Due to the advent setbacks of analogue transmission in signal frequency constriction it has been scrubbed off air. The High power capacity involved has also contributed to the same.

Figure 3.18: A typical KBC Analogue transmitter

The modulated signal is applied to a mixer (also known as frequency converter). Another input to the mixer which is usually produced in a crystal oven oscillator is known as subcarrier. The two outputs of the mixer are the sum and difference of two signals. Unwanted signal (usually the sum) is filtered out and the remaining signal is the RF signal.

Then the signal is applied to the amplifier stages. The number of series amplifiers depends on the required output power. The final stage is usually an amplifier consisting of many parallel power transistor systems. In some of the model transmitters tetrodes or klystrons are also utilized.

The Analogue system put about 70% to 90% of the transmitters power into the sync pulses. The remainder of the transmitter's power goes into transmitting the video's higher frequencies and the FM audio carrier.

The video signal modulates a carrier by a kind of amplitude modulation (VSB modulation or C3F). The modulation polarity is negative. That means that the higher the level of the video signal the lower the power of the RF signal. Channel 23 was in use with a frequency of 487.25 MHz and 10 kW power consumption.

3.10: Digital TV Transmission

This is the transmission of audio and video by digitally processed and multiplexed signal, in contrast to the totally analog and channel separated signals used by analog television. Digital TV can support more than one program in the same channel bandwidth.

Digital Video Broadcasting (DVB) used coded orthogonal frequency-division multiplexing (OFDM) modulation and supports hierarchical transmission.

Figure 3.19: KBC Digital TV Transmitter

3.10.1: Exciter (Channel 37)

An exciter is basically is a low powered transmitter (up to a couple hundred watts), which drives a power amplifier system (which would output several thousand watts or more).

The exciter generates the carried frequency (digitally generated on modern transmitters, crystal generated on old ones), and modulate it with the incoming audio (usually composite, sometimes discrete left/right, or even digital on modern transmitters), and output an RF carrier, which can directly drive an antenna, to the output power of the exciter.

The audio signal fed to the exciter is led to the HPB-1212 Stereo Modulator card or HPB-1213 mono operation for conversion into a stereo composite signal or monaural signal. The signal is fed through U-link located on the front panel and it is combined with the auxiliary signal through the HPB-1210 mother board and fed to the FM modulator.

This FM modulator uses the DCFM (Direct carrier frequency modulation) system to emit an RF signal of carrier frequency up to 10mW.

The HPB-1211 FM TR PA amplifies the RF signal from 2W to 20W when it is forced air cooling or 15W when it is convection cooling as output of the FM exciter.

The synthesized PLL (Phase locked loop) circuit for carrier frequency regulation, which is located on the mother board, divides the RF sample from the FM modulator into I/N, compares with the high stable crystal oscillator and feeds back the difference between them into the FM modulator. A value of N can be preset with the switches and it can be preset for a maximum 8channel with the HPB-1215 channel selector board (option) to make n+1 standby configuration

3.10.2: Filter

Digital filters are used for two general purposes:

(1) Separation of signals that have been combined needed when a signal has been contaminated with interference, noise, or other signals.

(2) Restoration of signals that have been distorted in some way.

Analog filters can be used for these same tasks; however, digital filters are seen to achieve far superior results.

Note: Analog filters are cheap, fast, and have a large dynamic range in both amplitude and frequency. Digital filters, in comparison, are vastly superior in the level of performance that can be achieved.

The digital filter has an impulse response, a step response and a frequency response containing complete information about the filter, but in a different form. If one of the three is specified, the other two are fixed and can be directly calculated describing how the filter will react under different circumstances.

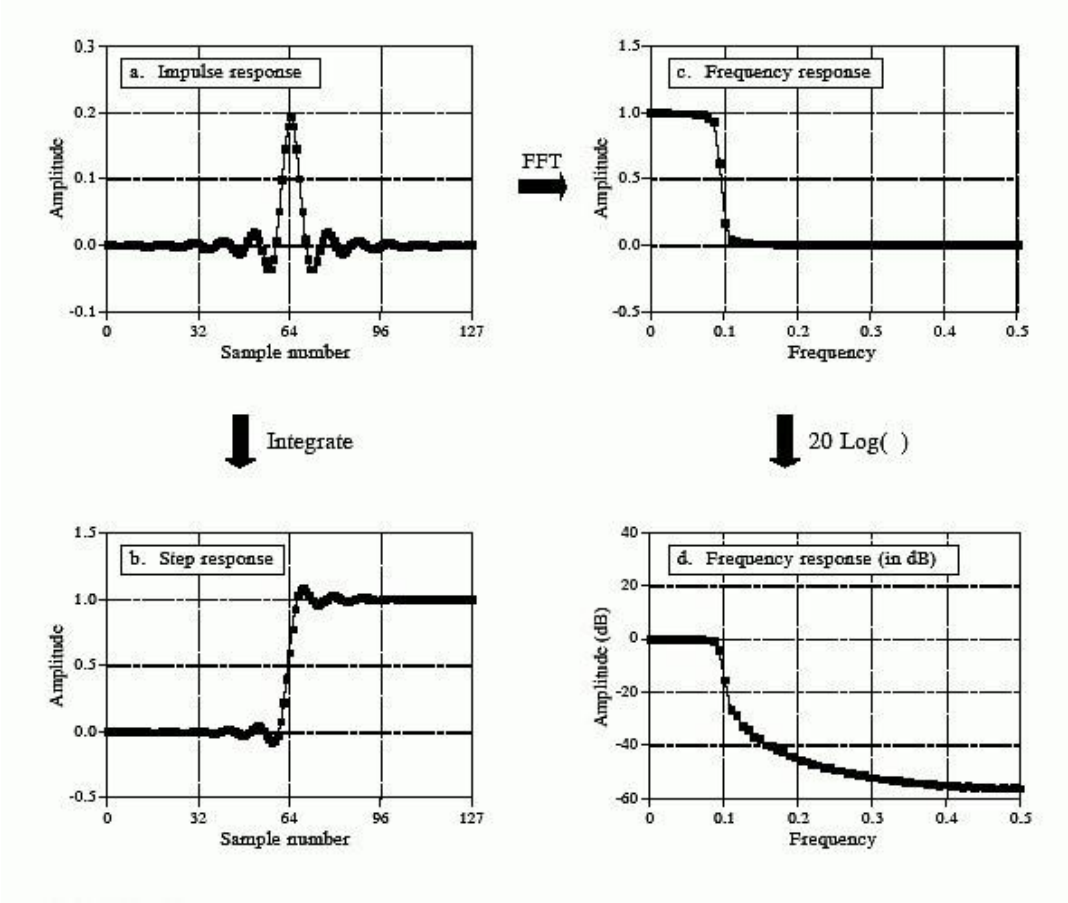


Figure 3.20: Digital Filter

The most straightforward way to implement a digital filter is by convolving the input signal with the digital filter's impulse response. Recursion method could also be used.

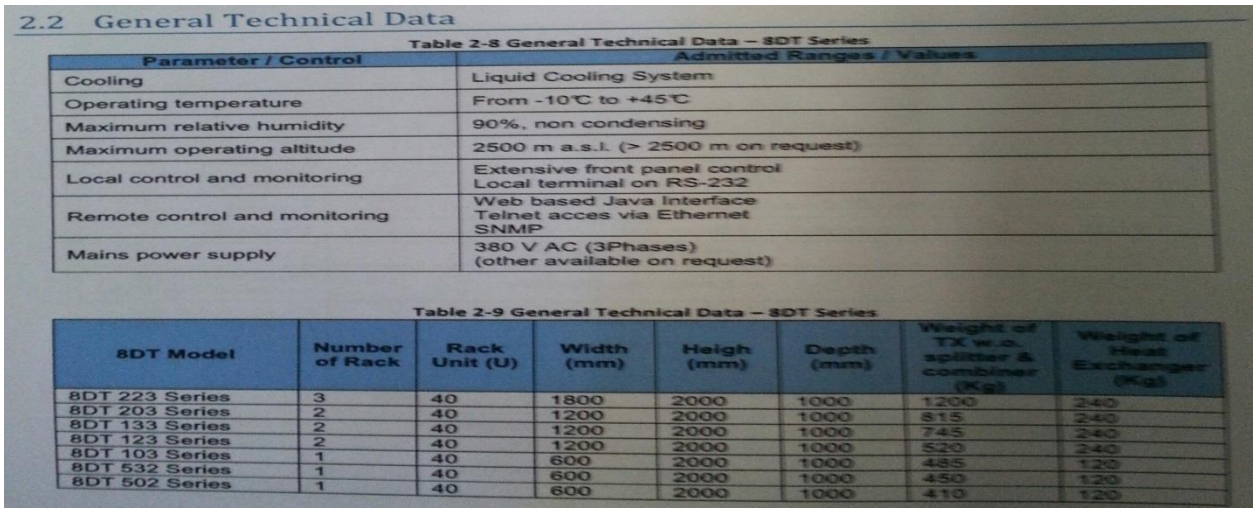
The power is seen to drop to 4.207 kW from 10 kW in the system set up.

3.10.3: Water cooling system

The signal is then propagated to a pumping water cooling system to counter on the dissipated heat in the process.

The main utility of the cooling liquid is that the heat produced by the transmitter is transported outside the plant and is dissipated by the radiator on the external environment, resulting in large savings in air conditioning

The system uses two pumps, one in reserve passive, ie, in case the pump is working fails, automatically activates the second. The hydraulic system that allows this change is entirely mechanical in the outlet valves of the pump allowing liquid to flow only in one direction. In this way, simply turn on and off the pumps for the hydraulic circuit automatically adapts.



2.2 General Technical Data

Table 2-8 General Technical Data – 8DT Series

Parameter / Control	Admitted Ranges / Values
Cooling	Liquid Cooling System
Operating temperature	From -10°C to +45°C
Maximum relative humidity	90%, non condensing
Maximum operating altitude	2500 m a.s.l. (> 2500 m on request)
Local control and monitoring	Extensive front panel control Local terminal on RS-232
Remote control and monitoring	Web based Java Interface Telnet acces via Ethernet SNMP
Mains power supply	380 V AC (3Phases) (other available on request)

Table 2-9 General Technical Data – 8DT Series

8DT Model	Number of Rack	Rack Unit (U)	Width (mm)	Heigh (mm)	Depth (mm)	Weight of TX w.o. splitter & combiner (Kg)	Weight of Heat Exchanger (Kg)
8DT 223 Series	3	40	1800	2000	1000	1200	240
8DT 203 Series	2	40	1200	2000	1000	815	240
8DT 133 Series	2	40	1200	2000	1000	745	240
8DT 123 Series	2	40	1200	2000	1000	520	240
8DT 103 Series	1	40	600	2000	1000	485	120
8DT 532 Series	1	40	600	2000	1000	450	120
8DT 502 Series	1	40	600	2000	1000	410	120

Figure 3.21: Technical data for respective parameters

NB: For channel 26 the system yields 4.139 kW and a combiner is involved.

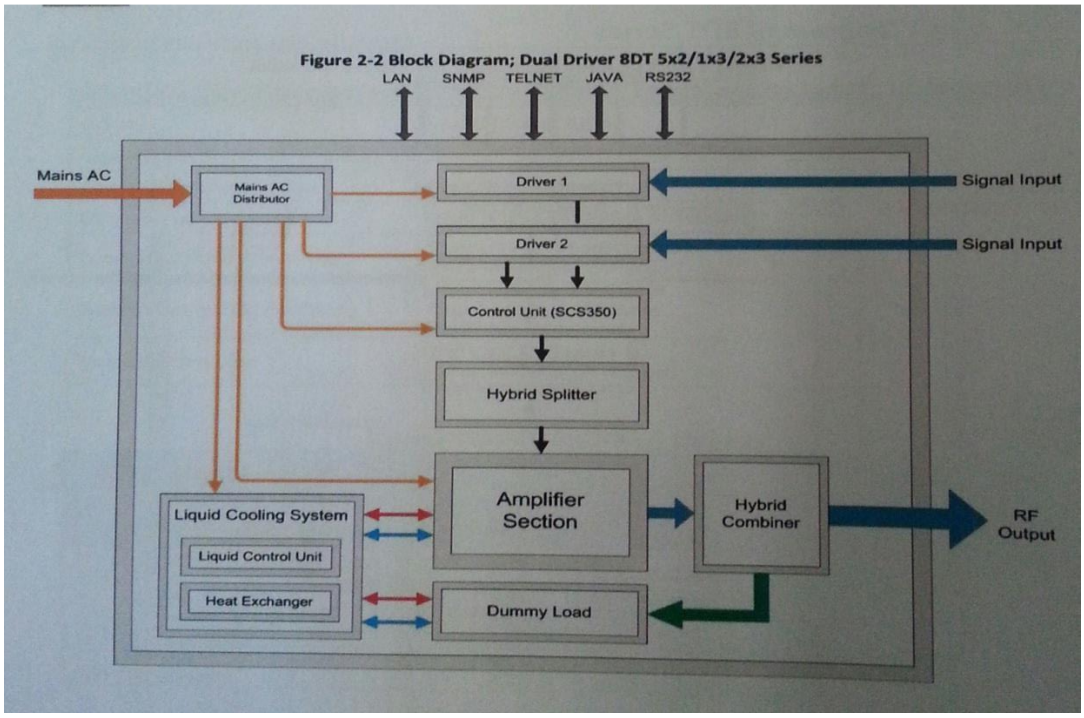


Figure 3.22: Block diagram for the dual driver

3.10.4: Testing

Testing is done via a dummy load where dummy antenna is connected to the output of the transmitter and electrically simulates an antenna, to allow the transmitter to be adjusted and tested without radiating waves. In testing the audio, a dummy load is connected to the output of the amplifier to electrically simulate a loudspeaker, allowing the amplifier to be tested without producing sound.

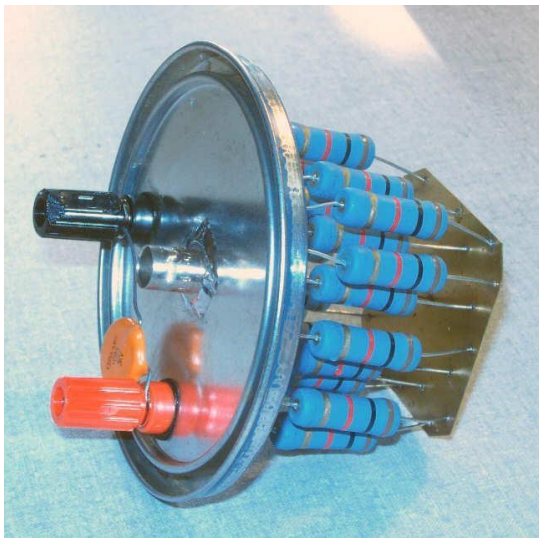


Figure 3.23: Dummy load model

When testing audio amplifiers the loudspeaker is replaced with a dummy load, so that the amplifier's handling of large power levels can be tested without actually producing intense sound. The simplest is a resistor bank to simulate the voice coil's resistance.

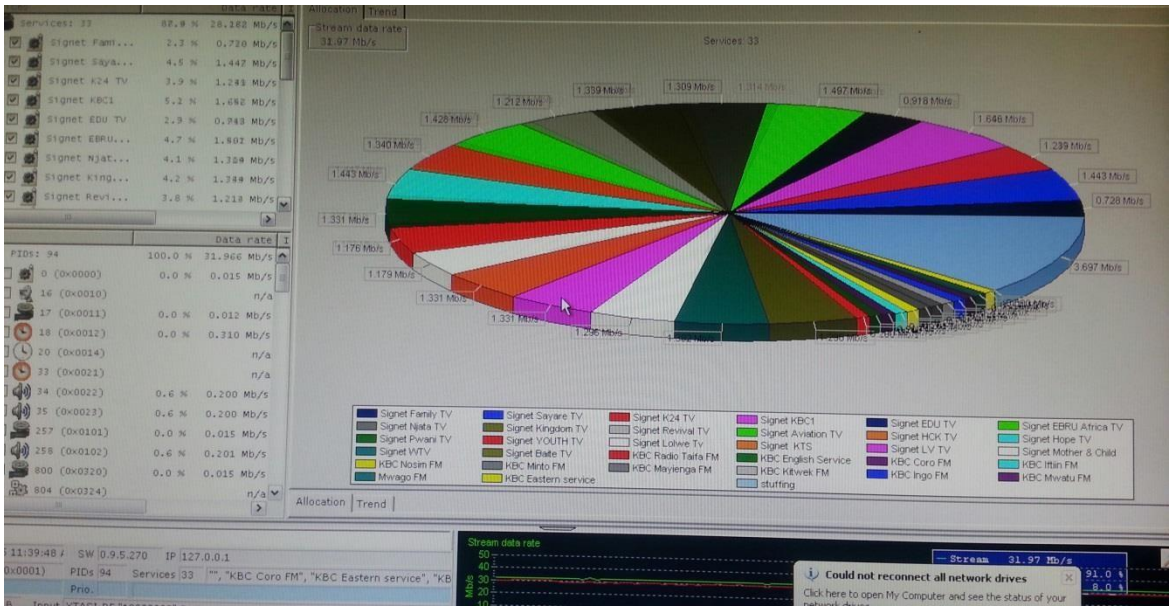


Figure 3.24: Pie chart display of the signal content

3.10.5: Demodulation

The original information-bearing signal is extracted from a modulated carrier wave by use of a 3 set demodulator outputting a digital signal.

The signal is fed into a Phase Locked Loop and the error signal is used as the demodulated signal.

3.10.6: Transmission

The signal can then finally be transmitted to the users where the receiver end would decode the content to obtain the specific frequencies on use.

Two methods are used depending on the signal content and mode of airing:

a) Microwave (point to point)
b) Satellite
c) Fibre optic

For instance parliament sessions are aired via Fibre optic means while normal programs are propagated through Microwave and long distance transmission is aired via satellite.

Figure 3.25: A typical transmitter

3.10.7: Decoding

On the receivers end the consumer gets to decode the obtained signal i.e. undoing the encoding so that the original information can be retrieved through a decoder to view the respective frequencies separately from the single stream.

CHAPTER 4

DISCUSSION

4.1: Analogue transmission

Analog signals were seen to be continuous signal which represents physical measurements denoted by sine waves whereas Digital signals are discrete time signals generated by digital modulation denoted by square waves.

The analog transmission lacked complex multiplexing procedures and timing equipment operating at high power capacities due to the continuity nature involved.

It formed a wider broadcast area i.e. Could travel over long distances due to the continuous trait. However, in situations where a signal often has high signal-to-noise ratio and cannot achieve source linearity, or in long distance, high output systems, analog is unattractive due to attenuation problems.

The main advantage is the fine definition of the analog signal which has the potential for an infinite amount of signal resolution. Compared to digital signals, analog signals are of higher density. Another advantage with analog signals is that their processing may be achieved more simply than with the digital equivalent. An analog signal may be processed directly by analog components, though some processes aren't available except in digital form.

The primary disadvantage of analog signaling is that any system has noise – i.e., random unwanted variation. As the signal is copied and re-copied, or transmitted over long distances, these apparently random variations become dominant.

Electrically, these losses can be diminished by shielding, good connections, and several cable types such as coaxial or twisted pair. The effects of noise create signal loss and distortion. This is impossible to recover, since amplifying the signal to recover attenuated parts of the signal amplifies the noise (distortion/interference) as well. Even if the resolution of an analog signal is higher than a comparable digital signal, the difference can be overshadowed by the noise in the signal.

4.2: Digital Transmission

Digital transmission was described as a method of storing, processing and transmitting information through the use of distinct electronic or optical pulses that represent the binary digits 0 and 1.
Digital transmission covered less area of broadcast in comparison to analog transmission due to the discrete nature. Transmission Boosters were laid along the line to propagate the signal to greater lengths.

4.2.1: Pros and cons

Advantages of Digital Transmission:

- Less expensive
- More reliable
- Easy to manipulate
- Flexible
- Compatibility with other digital systems
- Only digitized information can be transported through a noisy channel without degradation
- Integrated networks

Disadvantages of Digital Transmission:

- Sampling Error
- Digital communications require greater bandwidth than analogue to transmit the same information.
- The detection of digital signals requires the communications system to be synchronized, whereas generally speaking this is not the case with analogue systems.

Feature	Analog Characteristics	Digital Characteristics
1)Signal	Continuously variable, in both amplitude and frequency	Discrete signal, represented as either changes in voltage or changes in light levels
2)Traffic measurement	Hz (for example, a telephone channel is 4KHz)	Bits per second (for example, a T-1 line carries 1.544Mbps, and an E-1 line transports 2.048Mbps)
3)Bandwidth	Low bandwidth (4KHz), which means low data transmission rates (up to 33.6Kbps) because of limited channel bandwidth	High bandwidth that can support high-speed data and emerging applications that involve video and multimedia
4)Network capacity	Low; one conversation per telephone channel	High; multiplexers enable multiple conversations to share a communications channel and hence to achieve greater transmission efficiencies
5)Network manageability	Poor; a lot of labor is needed for network maintenance and control because dumb analog devices do not provide management information streams that allow the device to be remotely managed	Good; smart devices produce alerts, alarms, traffic statistics, and performance measurements, and technicians at a network control center (NCC) or network operations center (NOC) can remotely monitor and manage the various network elements
6)Power requirement	High because the signal contains a wide range of frequencies and amplitudes	Low because only two discrete signals—the one and the zero—need to be transmitted
7)Security	Poor; when you tap into an analog circuit, you hear the voice stream in its native form, and it is difficult to detect an intrusion	Good; encryption can be used
8)Error rates	High; 10–5 bits (that is, 1 in 100,000 bits) is guaranteed to have an error	Low; with twisted-pair, 10–7 (that, is 1 in 10 million bits per second) will have an error, with satellite, 10–9 (that is, 1 in 1 billion per second) will have an error, and with fiber, 10–11 (that is only 1 in 10 trillion bits per second) will have an error

9)Signal	Analog signal is a continuous signal which represents physical measurements.	Digital signals are discrete time signals generated by digital modulation.
10)Waves	Denoted by sine waves	Denoted by square waves
11)Representation	Uses continuous range of values to represent information	Uses discrete or discontinuous values to represent information
12)Technology	Analog technology records	Samples analog waveforms into
	waveforms as they are.	a limited set of numbers and records them.
13)Data transmissions	Subjected to deterioration by noise during transmission and write/read cycle.	Can be noise-immune without deterioration during transmission and write/read cycle.
14)Response to Noise	More likely to get affected reducing accuracy	Less affected since noise response are analog in nature

CHAPTER 5

CONCLUSION AND RECOMMENDATIONS

5.1: Conclusion

In conclusion an analysis was undertaken to an analogue transmitter of 100 kW laying in points the modes of operations and its pros and cons involved.

A digital transmitter was then set on design from the preceding analysis and the distinctive features it contained to transmit frequencies in a more compact stream through multiplexing.

5.2: Recommendations

- Linearity improvement in analog transmission

A method and system for improving the linearity of an analog transmission in a multichannel fiber optic transmission system uses a power series correction derived from a non-information bearing portion of a received transmission.

A notch filter reduces the energy in a portion of a guard band between channels before transmission so that the energy in the notch-filtered portion of the received transmission is indicative of the non-linarites introduced by the analog transmission system.

This could in turn minimize noise intrusion upon transmission and yield a more reliable signal by decreasing the error rate.

- Boosting up security personnel at the transmission sites

The security levels at the transmission sites were compromised as access within was quite easy and this provided a vulnerable area for alterations and signal manipulation

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